MyPBX

User Manual

Version 1.0.1.1

Yeastar Technology Co., Ltd

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1 Introduction

MyPBX — Embedded Hybrid IP-PBX for Small Businesses

MyPBX is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations (2-100 users per site). MyPBX also offers a hybrid solution (a combination of VoIP applications using legacy telecom equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

1.1 Features

• IVR(Automated Attendant)	• Call Transfer
• Blind Transfer	• Call Waiting
• Call Detail Records(CDR)	• Caller ID
• Call Forward	• Do Not Disturb(DND)
• Call Parking	• Music On Hold
• Call Pickup	• Ring Group
• Call Recording	• Route by Caller ID
• Call Routing	• Three-way Calling
• Paging and Intercom	• MRI(MyPBX Recording Interface)

1.2 Hardware Specification

1.2.1 Exterior Appearance

1) Front Side

Figure 1-1 MyPBX Front Panel Picture

No.	Identifying
①	Green Light：Indicates the power connection is normal.
②	Green Light：Indicates the server system is in working order
③	Green Light：Indicates the internet interface0 is in use
④	Green Light：Indicates the internet interface1 is in use
⑤	Red Light：stands for FXO port Green Light：stands for FXS port LED Dual (Green) fast blink：indicates FXS port is ringing LED Dual (Red) fast blink：indicates an incoming call

2) Back Side

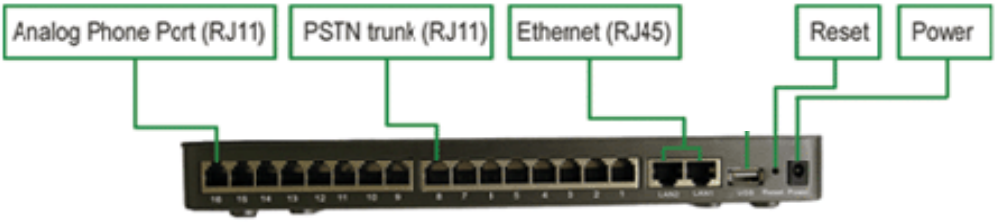


Figure 1-2 MyPBX Back Side Picture

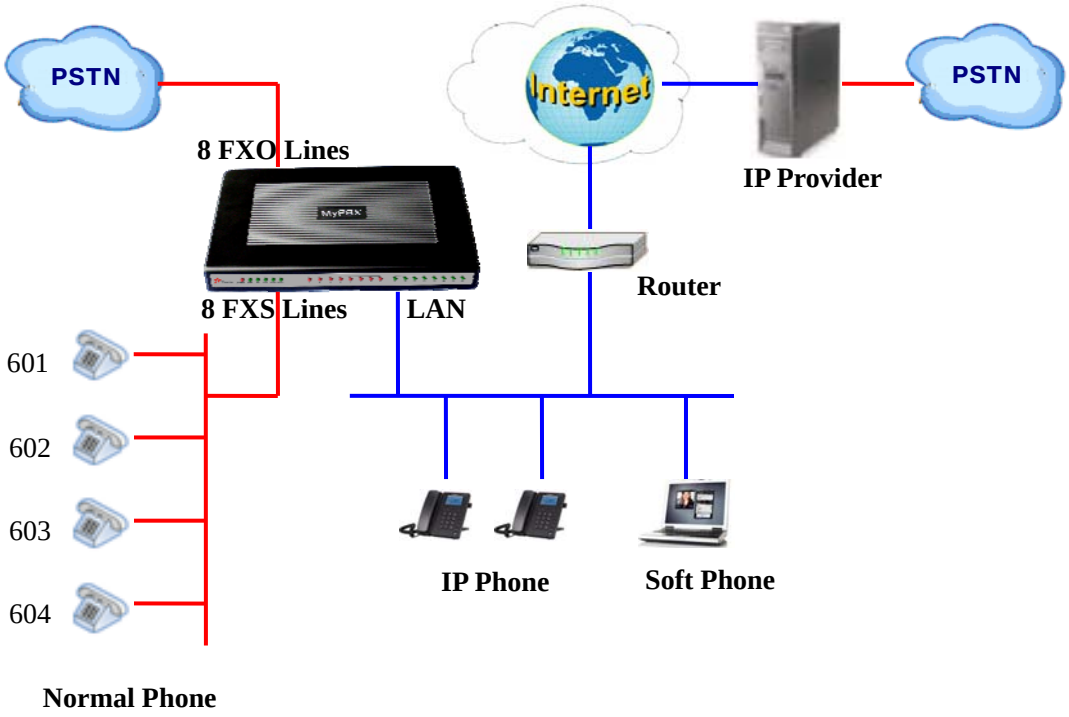
1.3 Package Contents

- 1 unit of MyPBX server
- 1 power supply
- 1 Internet line
- Few telephone cables [RJ-11]

Note: If any of the above items is damaged or missing, contact your reseller.

2 System set up

2.1 Connection Drawing



2.2 Connecting Ethernet Line

MyPBX provides two 10/100M Ethernet ports with RJ45 interface and LED. Besides transporting voice, Ethernet port can also send information of managing, maintaining and operating.
Plug Ethernet line into MyPBX's Ethernet port, and then connect the other head of Ethernet line with hub, switch, router, LAN or WAN. Check the status of LED after connection, yellow light indicates port is in connecting process, green light indicates the port is in working.

2.3 Connecting Power

MyPBX utilizes the high-performance switch power, which supply the enough voltage and electrical energy that required by MyPBX system.
AC Input: 100~240V
DC Output: 12V,5A

Note: Prior to connect power, it's better to use the three-phrase power outlet of neutral-point joint or multi-function computer power outlet.

Please follow the below steps to connect power socket:

1. Push the switch on 'OFF' on power outlet.
2. Connect one head of the accessory power cable with the power input port on MyPBX's back panel, and plug the other head of power cable into 220V power socket.
3. Push the switch on 'ON' on power outlet.
4. After wired up power, please check the Power LED on front panel whether it lighted or not. Lighted LED indicates the correct operation of power; if unlighted, please repeat the step 1 to 3 again.

3 Manage MyPBX

3.1 Manager Log in

Double click icon to open the IE browser, and input the IP address of MyPBX server.
If user is first time to configure MyPBX by web, please use the MyPBX default IP address to log in: http://192.168.5.150

The default Username is admin and password is password.

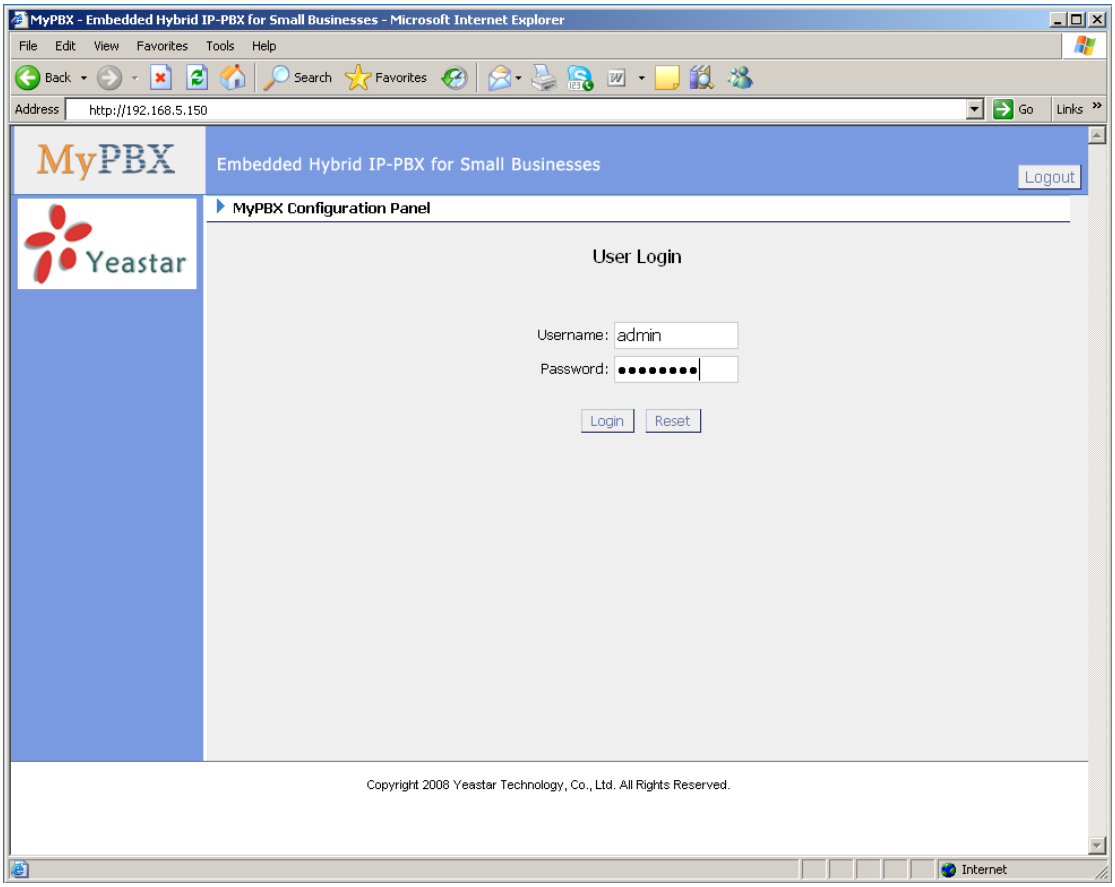


Figure 3-1

3.2 Status Monitor

3.2.1 Line Status

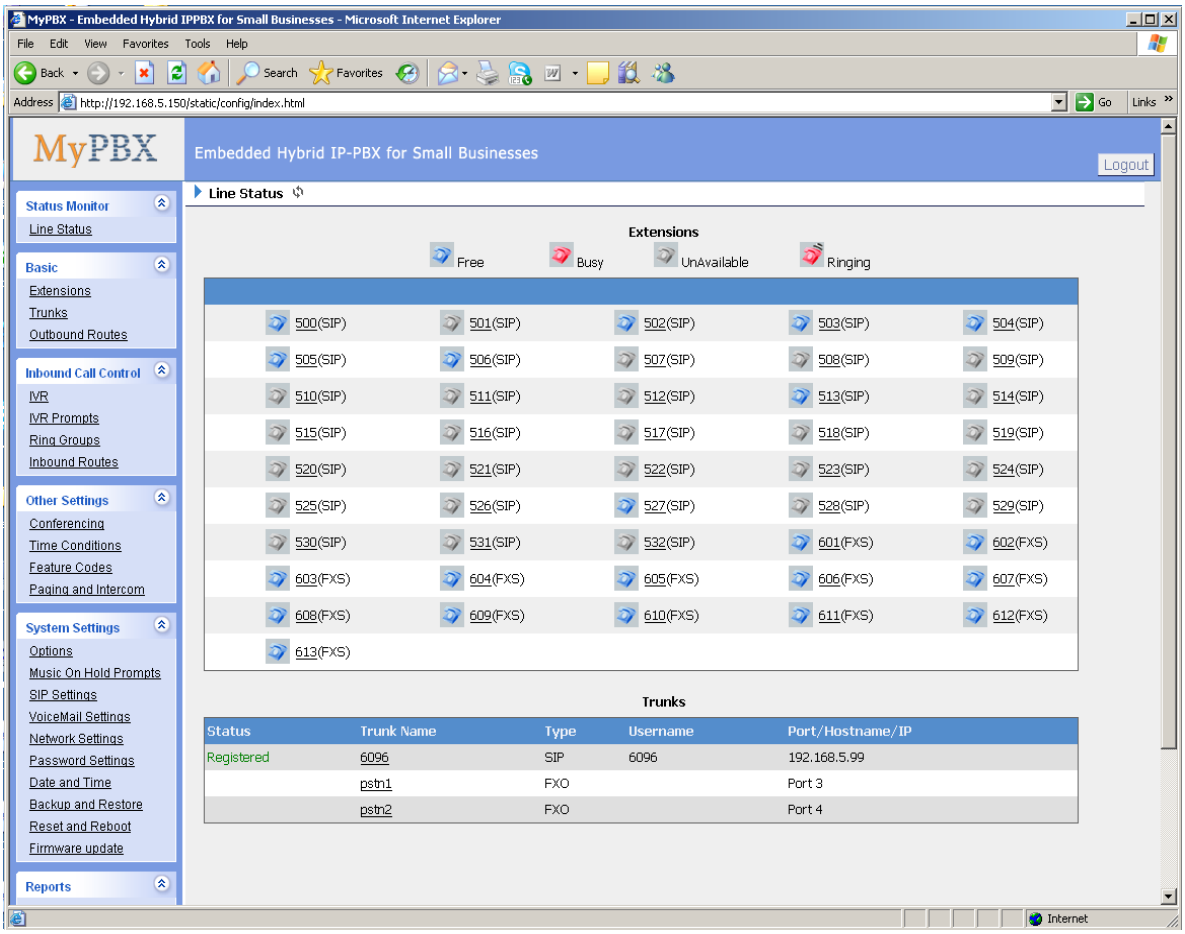


Figure 3-2

MyPBX Status Description:

Extensions:

1) : extension is unregistered

2) : extension is idle

3） : extension is ringing

4） : extension is busy in a call

Trunks:

SIP Trunk:

1) Unregistered — trunk registration failed.
2) Registered — succeed registration, trunk is ready to use.
3) Request Send — Registering.
4）Waiting — Waiting for Authentication.

FXO Trunk:
FXO: please refer to the LED identifying of front panel.

3.3 Basic

3.3.1 Extension

Extension has two types: normal desk extension (FXS) and SIP extension.

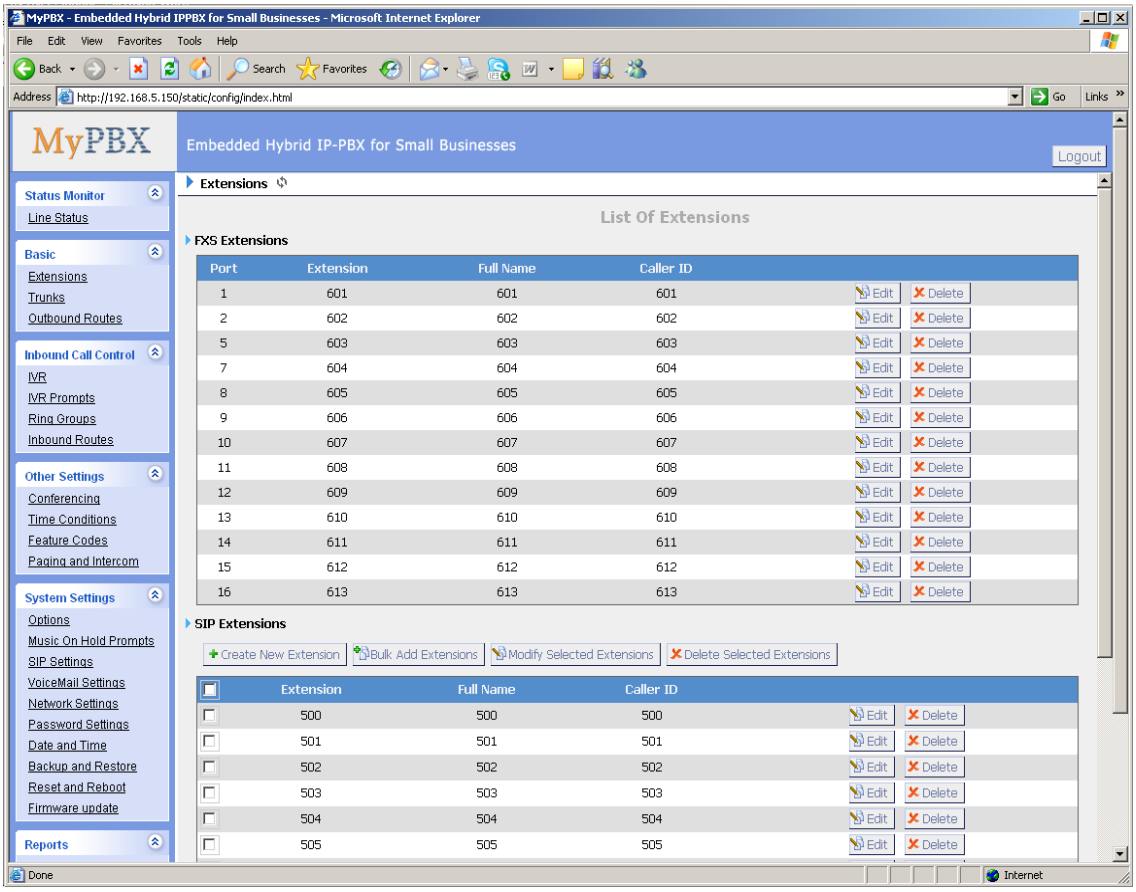


Figure 3-3

3.3.1.1 Normal Desk Phone (FXS）Extension

Edit Normal Desk Phone Extension

On the administration page of FXS extensions, click 'Edit' on the extension that you want to edit, and modify the following information on prompt window:

1) General

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Name

A character-based name for this user, i.e. 'Bob Jones'

·Caller ID

The Caller ID (CID) string would be used when this user calls another internal user.

2) Voice Mail

·Voice Mail Access PIN Code

Voicemail Password for this user, Ex: '1234'.

·Send Voice Mail to Email Address

This option defines whether or not voicemails are sent to the Email Address as attachments. Note: You need to have an smtp server configured for this functionality.

3) Flash

·rxflash

sets the amount of time, in milliseconds, that must have passed since the last hook-flash event received by MyPBX before it will recognize a second event. If a second event occurs in less time than defined for Flash, then MyPBX will ignore the event. The default value of Flash is 1000 ms, and it can be configured in 1ms increments.

4) Group

·Pickup Group

If a phone belongs in a pickup group that matches one of the call's pickup groups, that phone may pickup the incoming call by calling *4 on his phone(*4 is default setting, it can be changed in page Feature Codes -> General -> Call Pickup).

5) Follow me

The mode of call forwarding when there's incoming call on this extension. Admin can set up the call forwarding type for this extension.

6）Other Options

.Call Waiting

Check this option if the User or Phone should have Call-Waiting capability. If this option is checked, busy transfer will not be available.

.hasmanager

Check this option if allow user to access MRI(MyPBX Recording Interface:Check Voice Mail).Users may access MRI by using extension number as username and voicemail secret as password.

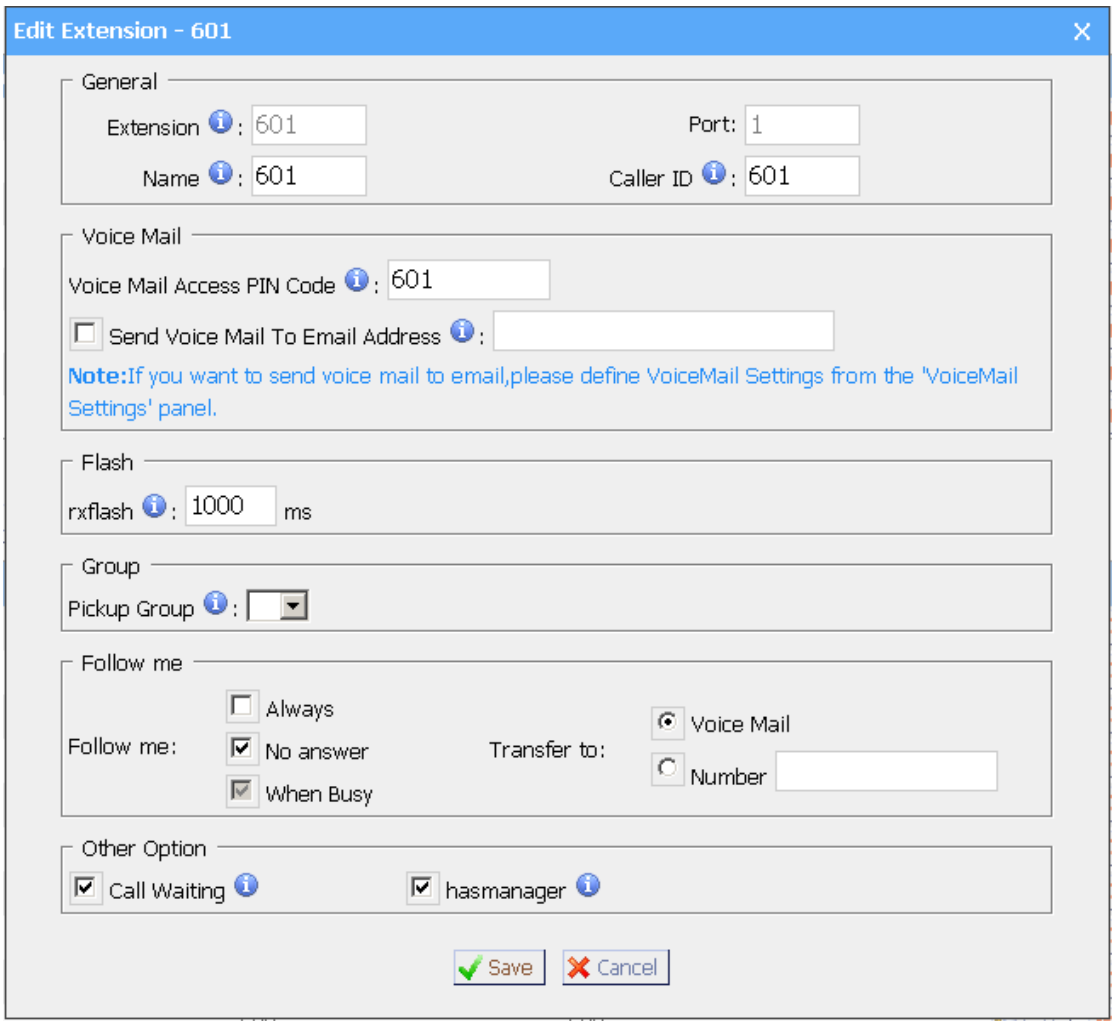


Figure 3-4

3.3.1.2 SIP Extension

SIP extension is an SIP Account allows your IP Phone or Soft IP Phone register to the MyPBX.

1. Add SIP Extension

Go to Extensions → SIP Extensions → Create New Extension

1) General

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Password

The password for the user's sip account , Ex: '12t3f6'

·Name

A character-based name for this user, i.e. 'Bob Jones'

·Caller ID

The Caller ID (CID) string would be used when this user calls another internal user.

2) Voice Mail

·Voice Mail Access PIN Code

Voicemail Password for this user, Ex: '1234'.

·Send Voice Mail to Email Address

This option defines whether or not voicemails are sent to the Email Address as attachments. Note: You need to have an smtp server configured for this functionality.

3) VoIP Setting

·NAT

Try this setting when MyPBX is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP ports.

·DTMF Mode – RFC2833，Info，Inband，Auto

4) Group

·Pickup Group

If a phone belongs in a pickup group that matches one of the call's pickup groups, that phone may pickup the incoming call by calling *4 on his phone(*4 is default setting,it can be changed in page Feature Codes>General>Call Pickup).

5) Follow me

The mode of call forwarding when there's incoming call on this extension. Admin can set up the call forwarding type for this extension.

6）Other Options

.Call Waiting

Check this option if the User or Phone should have Call-Waiting capability. If this option is checked, busy transfer will not be available.

.hasmanager

Check this option if allow user to access MRI(MyPBX Recording Interface:Check Voice Mail).Users may access MRI by using extension number and extension password.

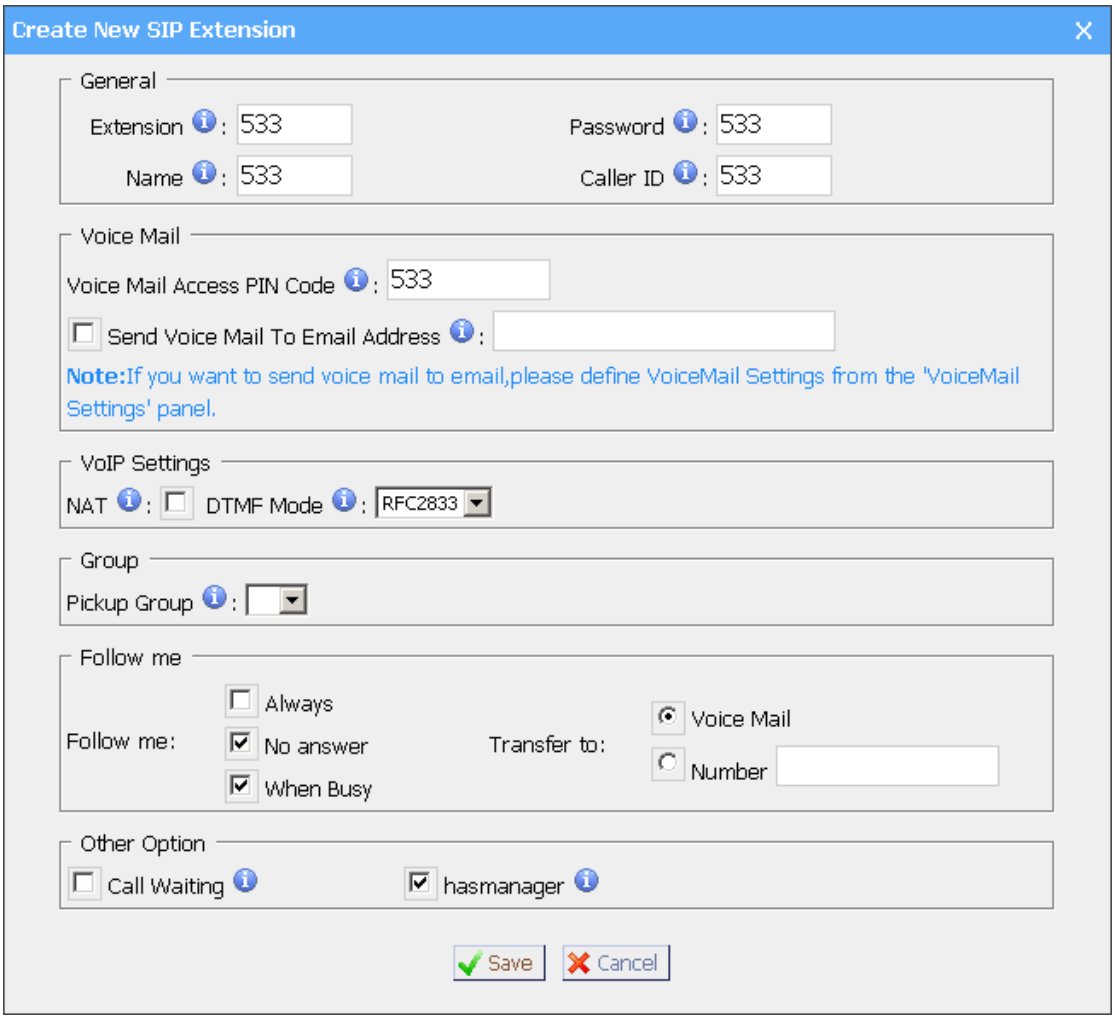

Create New SIP Extension
General
Extension : 533
Password : 533
Name : 533
Caller ID : 533
Voice Mail
Voice Mail Access PIN Code : 533
Send Voice Mail To Email Address :
Note:If you want to send voice mail to email,please define VoiceMail Settings from the 'VoiceMail Settings' panel.
VoIP Settings
NAT : DTMF Mode : RFC2833
Group
Pickup Group :
Follow me
Follow me: Always No answer When Busy
Transfer to: Voice Mail Number
Other Option
Call Waiting hasmanager
Save Cancel

Figure 3-5

2. Bulk add SIP Extensions

Go to Extensions → SIP Extensions → Bulk add Extensions

1）Choose the extension quantity you want to create

2）The initiative extension number when starting

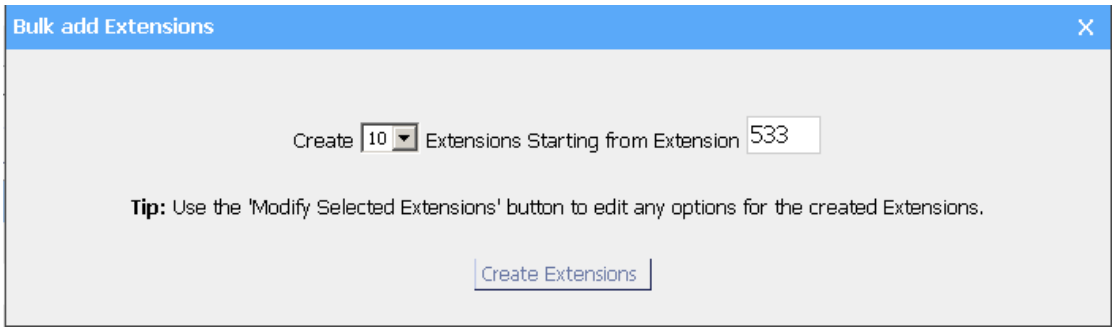

Bulk add Extensions
Create 10 Extensions Starting from Extension 533
Tip: Use the 'Modify Selected Extensions' button to edit any options for the created Extensions.
Create Extensions

Figure 3-6

3. Edit SIP Extension

Click 'Edit' on SIP Extension administration page or click 'Modify Selected Extensions' to edit.

3.3.2 Trunk

Trunk have two types: PSTN Trunk(FXO) and SIP Trunk.

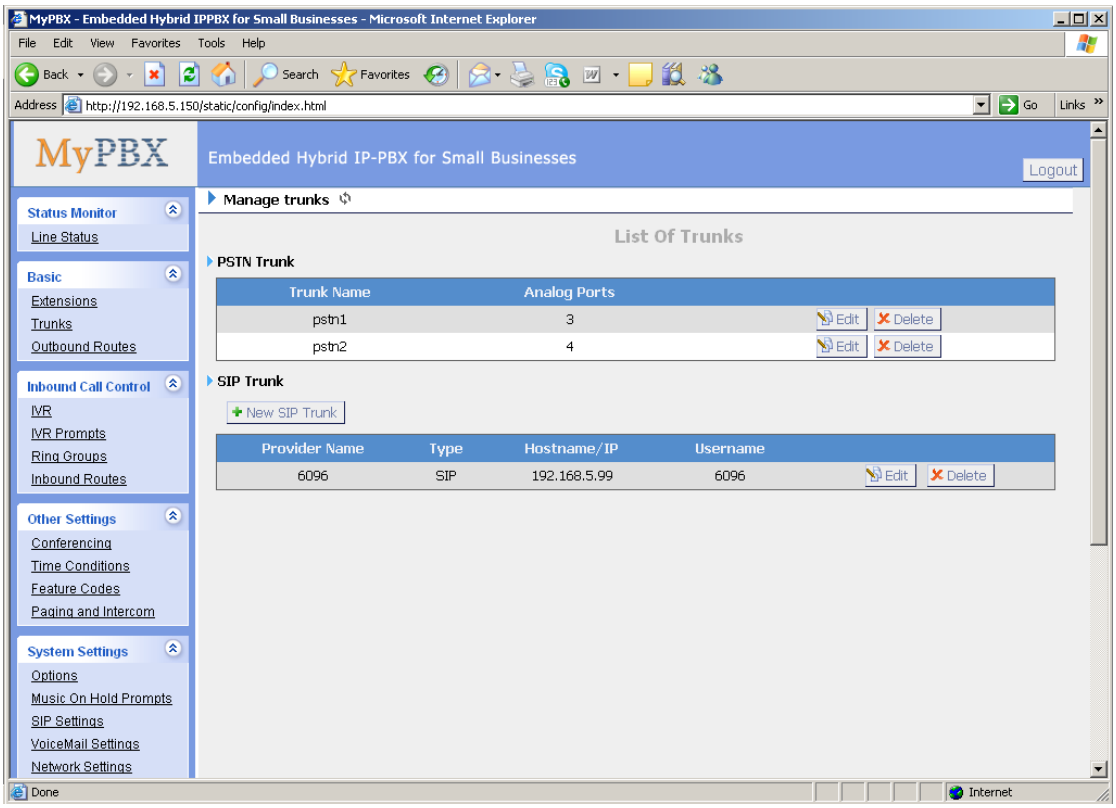


Figure 3-7

3.3.2.1 PSTN（FXO） Trunk

1. Edit PSTN（FXO） Trunk

On PSTN Trunk administration page, click 'Edit' on selected trunk and modify information on prompt window:

1）General

·Trunk Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'Port 5'

·Audio Setting

Normally you should not have to adjust your analog ports beyond the initial calibration.Should you still need to fine tune your audio settings, please use the adjustments.

2）Advanced Options

·Busy Detection

Busy Detection is used to detect far end hangup or for detecting busy signal. Enable to turn this feature on.

·Busy Count

If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before hanging up. The default is 4, but better results can be

achieved if set to 6 or even 8. Mind that the higher the number, the more time that will be needed to hangup a channel, but lowers the probability that you will get random hangups.

·Busy Pattern

If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal. In many countries, it is 500msec on, 500msec off. Without Busy Pattern specified, MyPBX will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then MyPBX will further check the length of the sound (tone) and silence, which will further reduce the chance of a false positive.

·Caller ID Start

This options allows one to define the start of a Caller ID signal:
Ring, to start when a ring is received (Caller ID Signaling: Bell_USA, DTMF).
Polarity, to start when a polarity reversal is started (Caller ID Signalling: V23_UK,V23_JP,DTMF).
Before Ring, to start before a ring is received (Caller ID Signaling: DTMF).

·Caller ID Signaling

This option defines the type of Caller ID signaling to use: bell (bell202 as used in the United States), v23_UK (as used in the UK), v23_JP (as used in Japan), or dtmf (as used in Denmark, Sweden, and Holland).

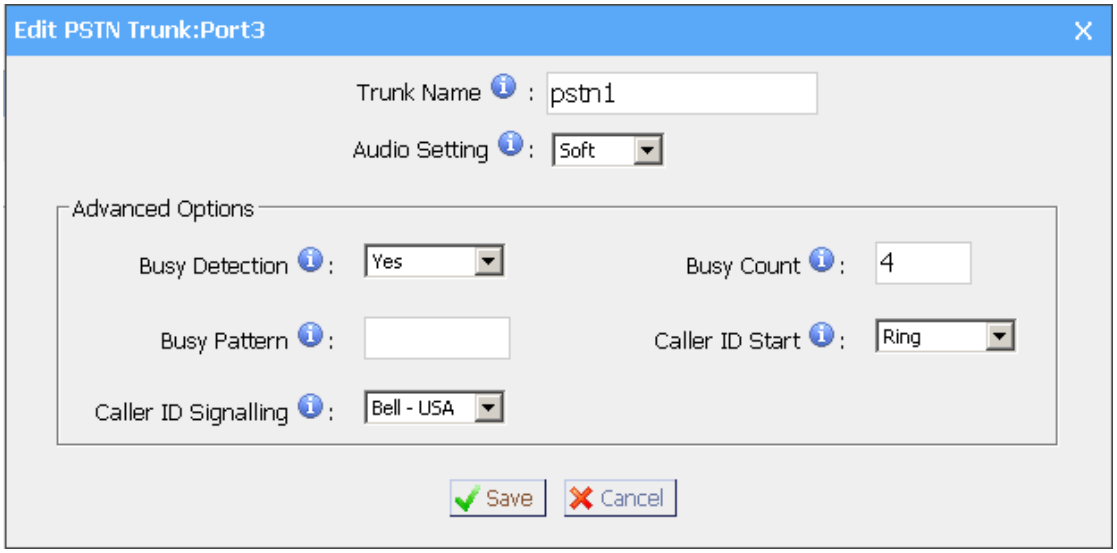


Figure 3-8

3.3.2.2 SIP Trunk

1. Add SIP Trunk

User must input correct SIP detailed information (that provided by VOIP provider), otherwise, SIP trunk can't work normally.

·Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'yeastar'.

·Hostname/IP

Service provider's hostname or IP address.
5060: standard port number used for SIP protocol, Don't change it if not required.

.Domain

VoIP provider's server domain name.

·Username

Username of SIP account. Used for SIP trunk registration.

.Authorization name

Used for SIP authentication. Leave this blank if not required.

·Password

Password of SIP account.

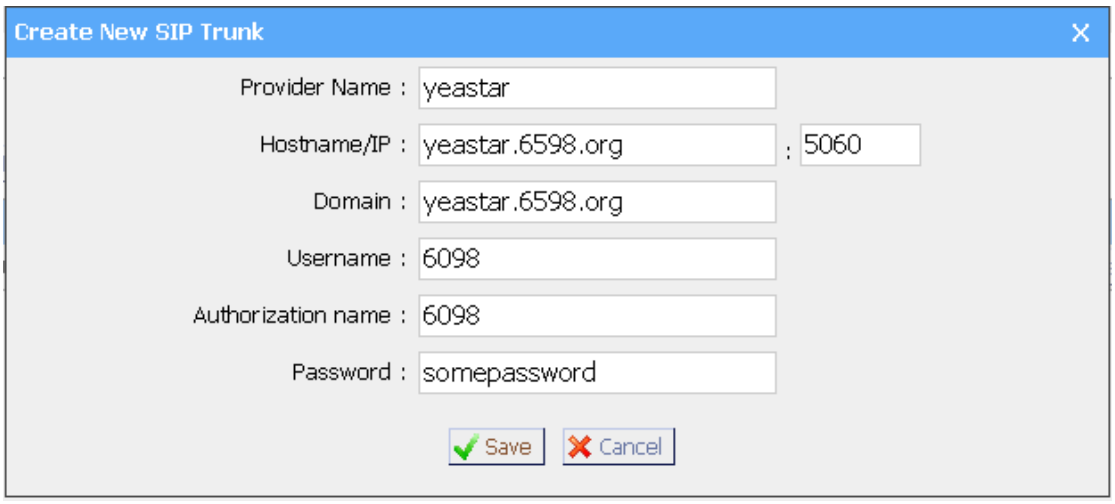


Figure 3-9

2. Edit SIP Trunk

Click 'Edit' on SIP trunk administration page to modify SIP trunk information.

3.3.3 Outbound Routes

Outbound routing mainly works for guides outgoing calls to go through trunks.

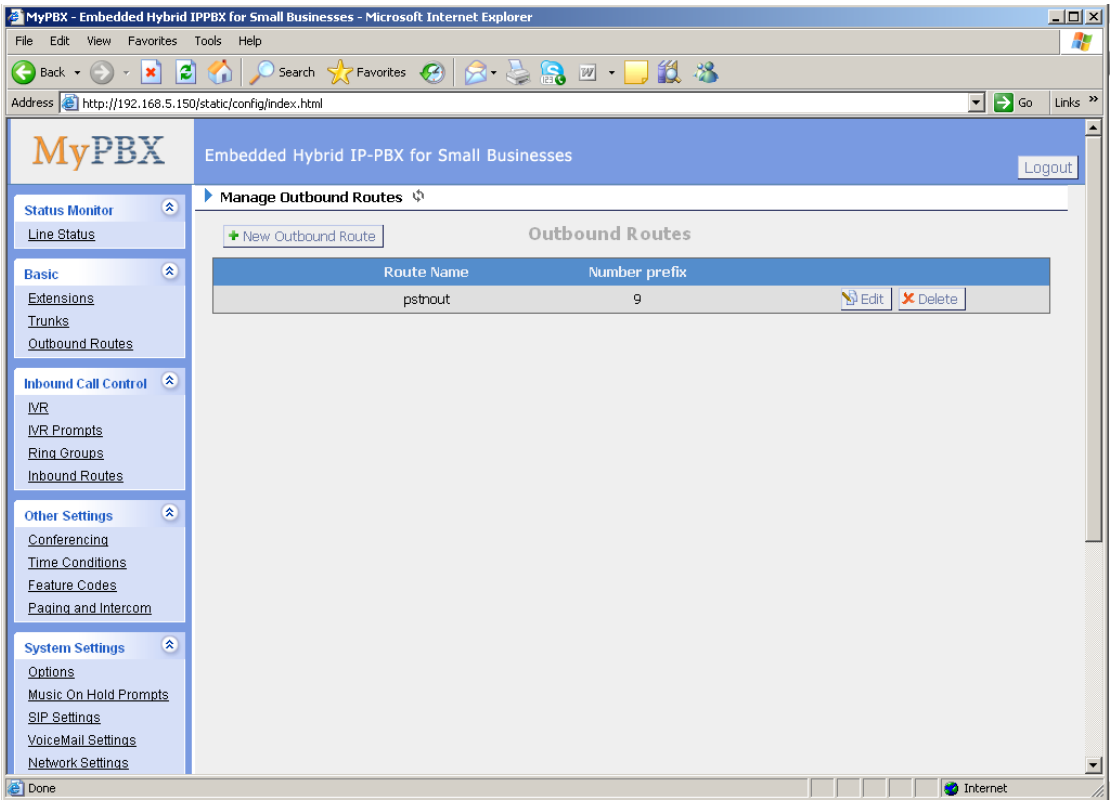


Figure 3-10

3.3.3.1. Create Outbound Route

Click 'Add Outboud Route' and fill in the responding information on prompt window.

1）General

·Route Name

Name of this Outbound Route. Ex: 'Local' or 'LongDistance' etc.

·Number Prefix

Calls to Numbers starting with

·Strip

Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in 'Use Trunk' One might; for example, want users to dial 9 before their long distance calls; however one does not dial 9 before those calls are placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.

·Prepend digits before dialing

Allows the user to specify digits that are prepend before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing

·Calls From Extensions

Defines the Extension that calls, matching the specified pattern, will be placed through.

·Make Outbound Calls on Trunk

Defines the Trunk that calls, matching the specified pattern, will be placed through.

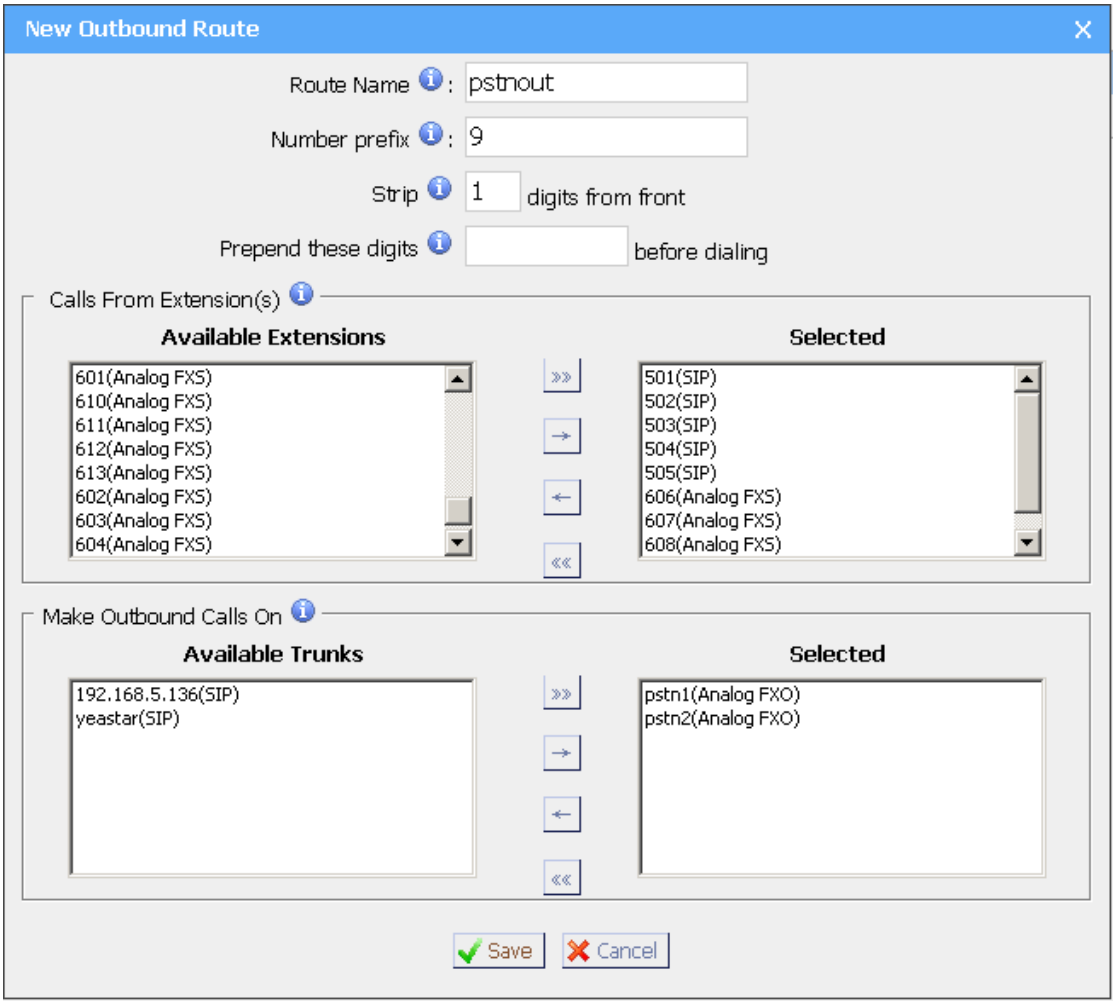


Figure 3-11

3.3.3.2. Delete Outbuond Route

Click 'Delete' on selected trunk to remove the trunk.

3.4 Inbound Call Control

3.4.1 IVR

When there's an incoming call aims at Auto Attendant, MyPBX will play the IVR and guide the caller to the required extensions (such as 'Welcome to XX company, for sales press 1, for technical support press 2, for operation press 0, or dial ext. number directly. System will transfer the call to corresponding extension according to DTMF).

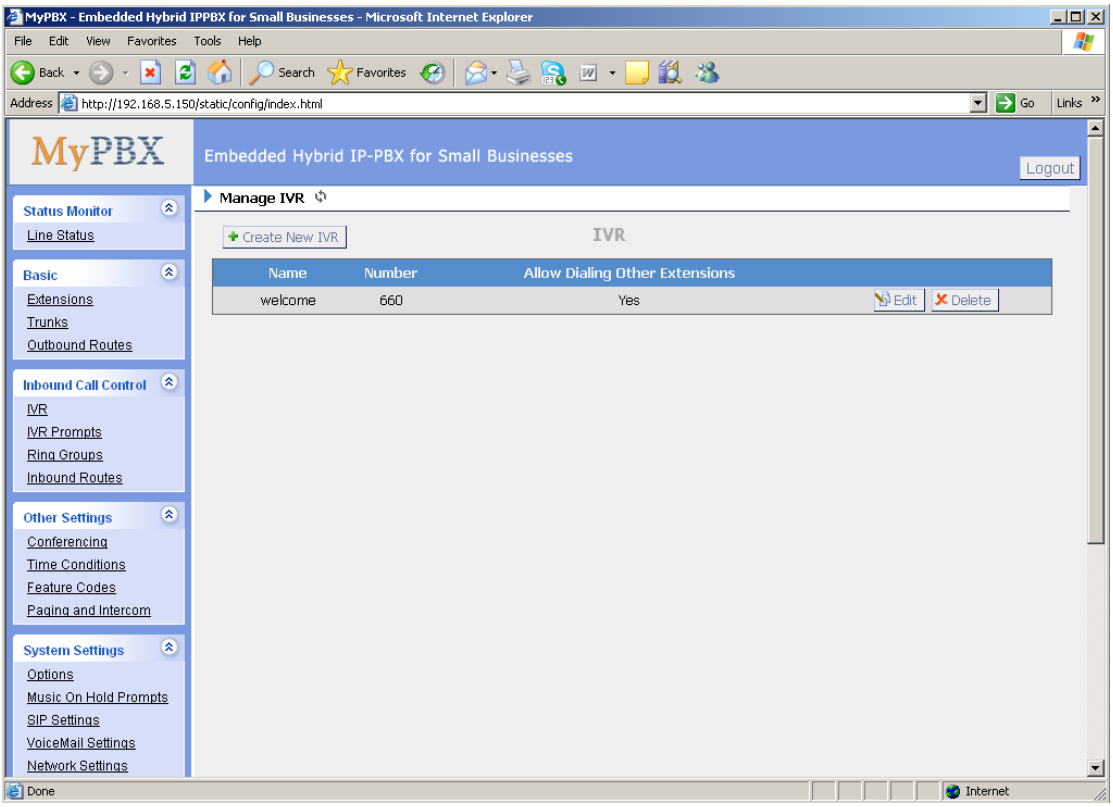


Figure 3-12

3.4.1.1 Auto Attendant

Click 'Create New IVR' as shown on Figure 3-13.

1）General

·Number

My PBX treats IVR as an extension; you can dial this number access the IVR.

·Name

A name for the IVR

·Prompt

A Prompt file for the IVR

·Allow Dialing Other Extensions

Check this option if the user want to dial extensions directly.

2）Key Press Events

Allow key press events will cause the system to listen for DTMF input from the caller and define the actions that occur when a user presses the corresponding digit.

·Time Out

Defines the timeout action. Timeout occurs when no DTMF entry is detected for 3 times after the IVR has finished playing its prompts.

·Invalid

Defines the invalid action. The invalid action is triggered if the user enters a DTMF that is not otherwise defined for the IVR.

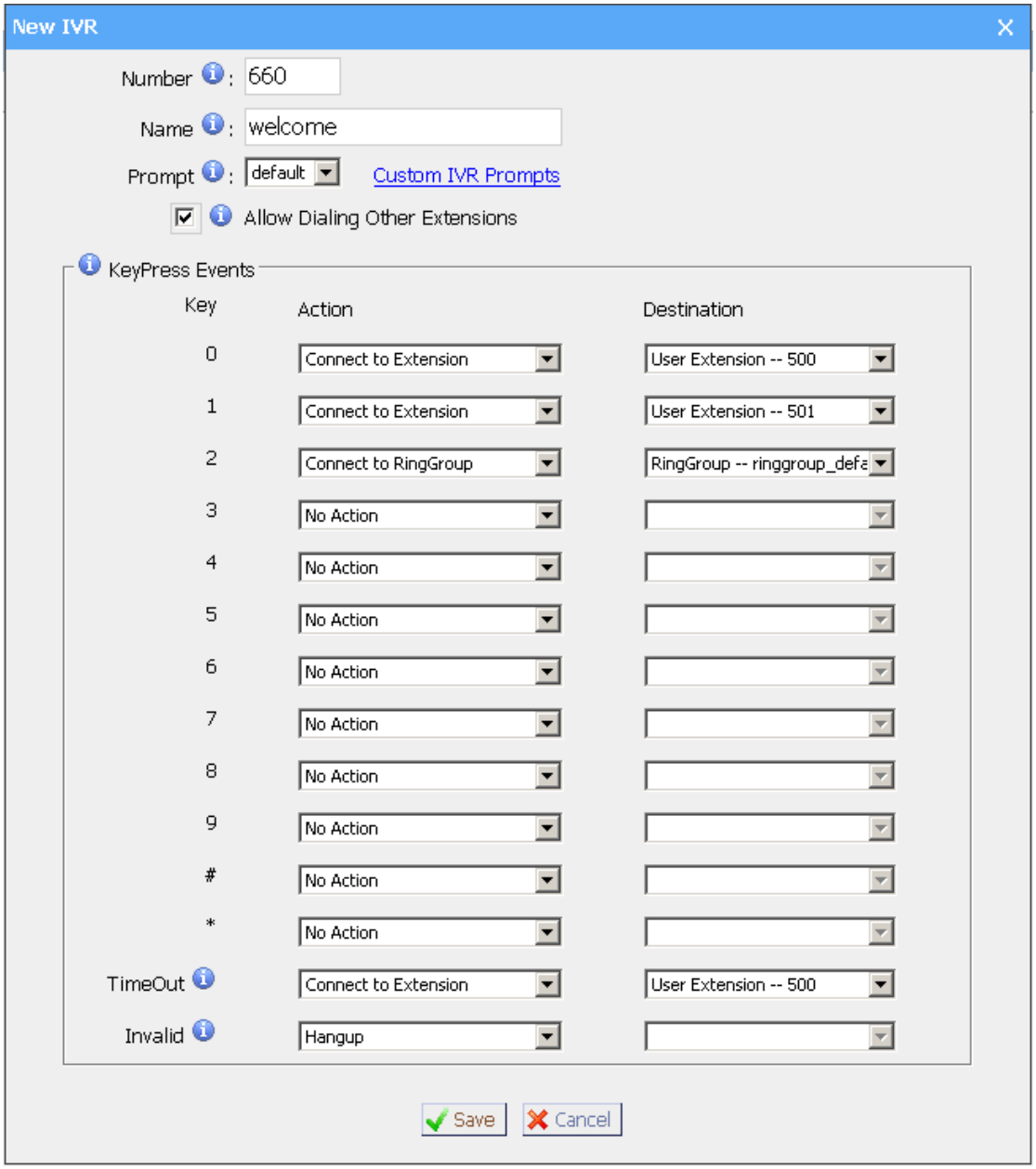


Figure 3-13

3.4.1.2 Delete IVR

Select the item you'd like to remove and click 'Delete' key.

3.4.2 IVR Prompts

Admin can record the IVR by himself according to his own requires:
1）Click 'Record New IVR Prompt'
2) Input File Name on prompt windows and choose an extension for recording such as 500.
3) Click 'Record', selected extension will ring and you can pick up the phone to start recording

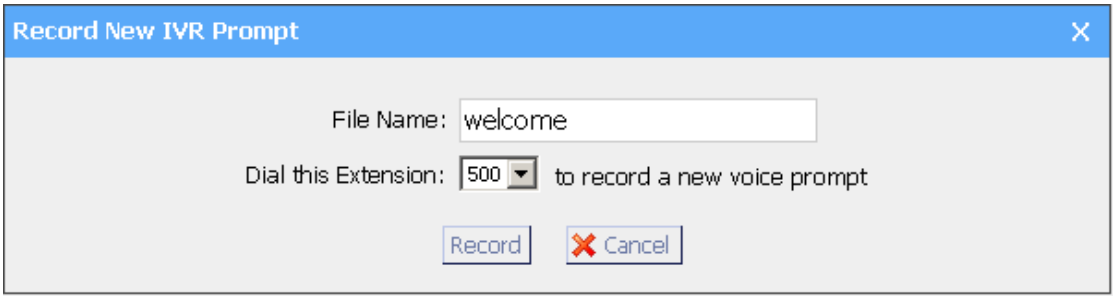


Figure 3-14

Admin can also upload IVR prompts:
1）Click 'Upload a IVR Prompt'.
2) Click 'Browse' to choose the IVR Prompt.
3) Click 'Upload' to upload the selected Prompt.

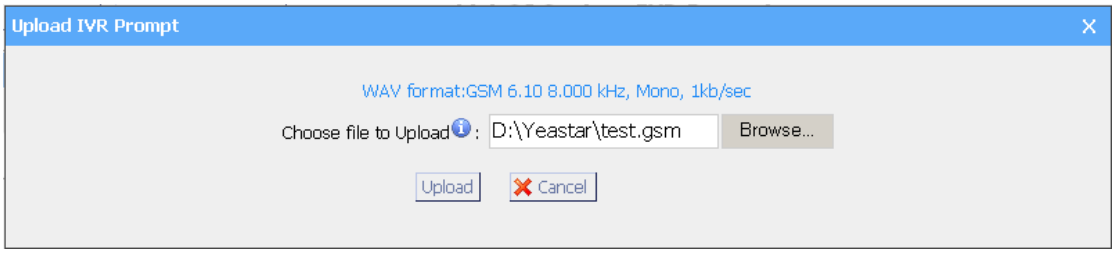


Figure 3-15

3.4.3 Ring Groups

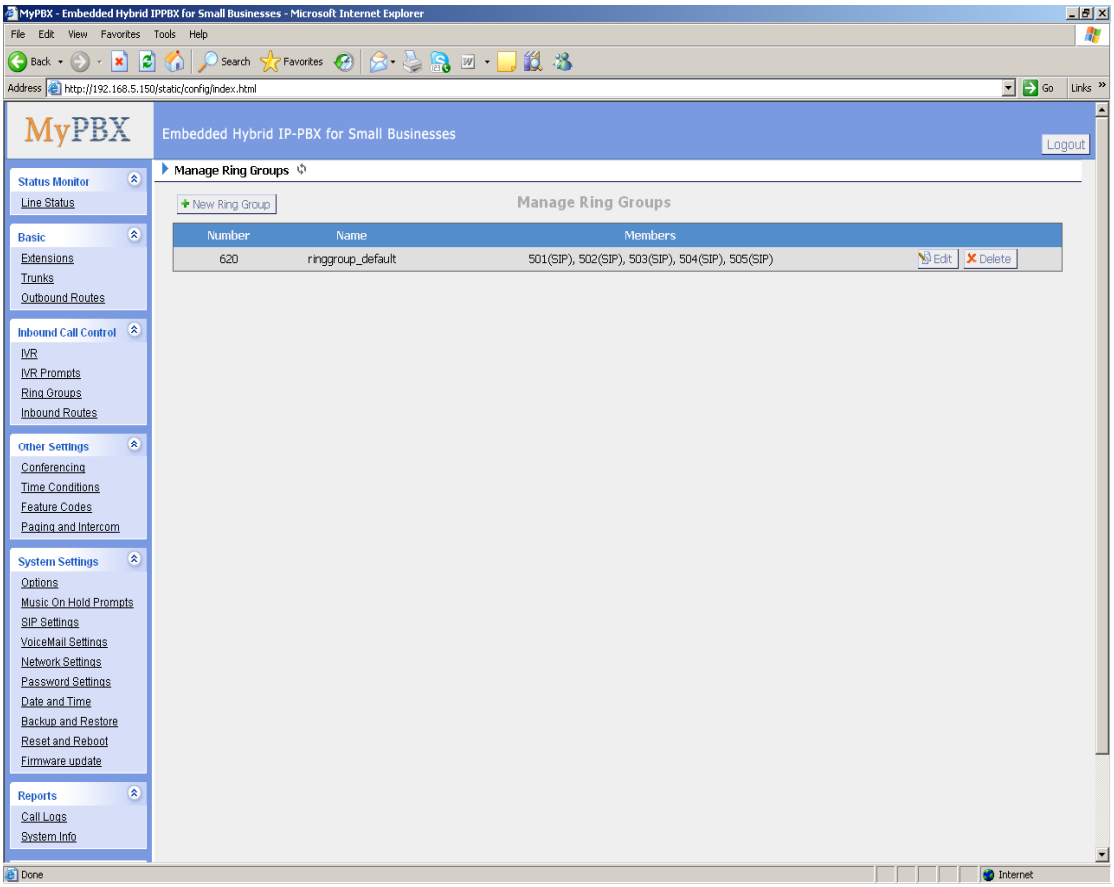


Figure 3-16

3.4.3.1 Create Ring Group

Click 'New Ring Group' to enter into the page

1）General

·Ring Group Name

This option defines a name for this Group, i.e. 'Sales'. 'Ring Group Name' is a label to help you see this group in the group list.

·Number for this ring group

This option defines the numbered extension that may be dialed to reach this Group.

·Strategy

This option sets the Ringing Strategy for this Group. The options are

1. Ring All: Ring All available Extensions simultaneously until one answers.
2. Ring in order: Take turns ringing each available Extension

·Time Out / Seconds to ring each member

1) Seconds to ring each member

How many seconds an Extension's phone will ring before MyPBX tries to ring the next Extension.

2) Time out

How many seconds the Extension's phone will ring.

·Ring Group Members

This selection shows all Users. Checking a User here makes them a member of the current Group.

·Destination If No Answer

When all people on this group are failed to answer the call, system will handle in this way.

Note：Welcome music of this group can be changed on System Settings → Options → Music On Hold

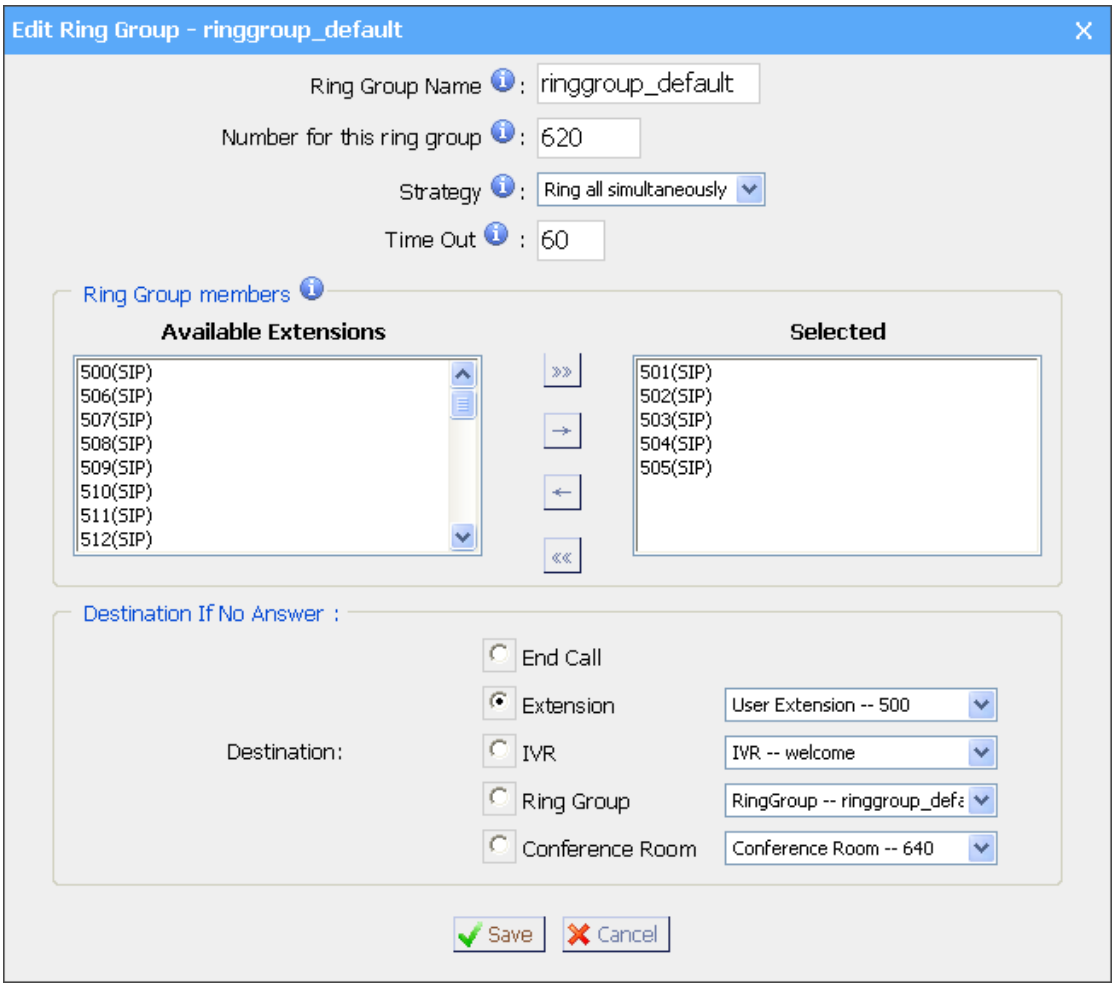


Figure 3-17

3.4.3.2 Delete Ring Group

Select the item you'd like to remove and click 'Delete' key.

3.4.4 Inbound Routes

Inbound routing mainly works for guides incoming calls to reach destination extension within the business hours or out of working hours.

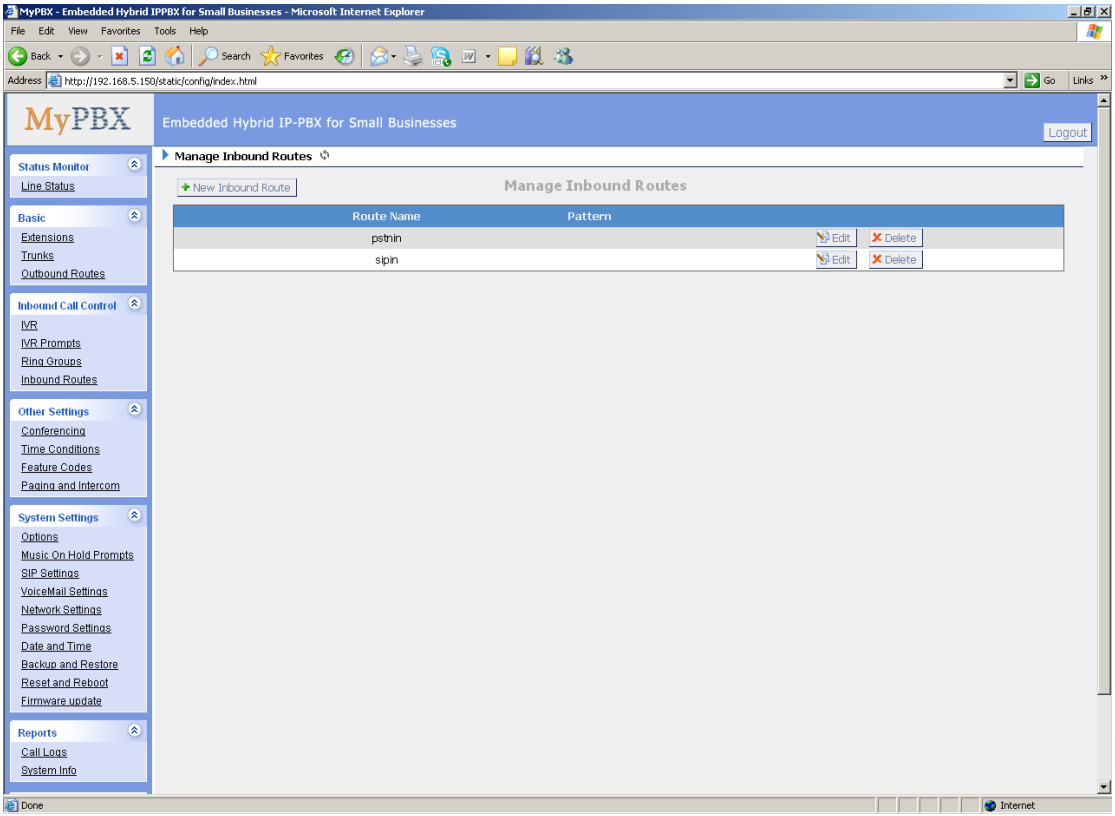


Figure 3-18

3.4.4.1 Create Inbound Route

Click 'New Inbound Route' to enter to the page.

1）General

·Route Name

Name of this Inbound Route. Ex: 'pstncallin' etc.

·Pattern

Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no CID info.

In patterns, some characters have special meanings:

X ... Any Digit from 0-9

Z ... Any Digit from 1-9

N ... Any Digit from 2-9

[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

. ... Wildcard, Matches Anything Remaining; i.e. 9011. Matches anything starting with 9011 (excluding 9011 itself)

! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

For example, **NXXXXXX** would match normal 7 digit dialings, while **1NXXNXXXXX** would represent a three digit area code plus phone number, proceeded by a one.

2）Trunk Sequence

This selection shows all Trunks. Checking Trunk here makes them a member of

the Inbound Route.

3）During Office Hours

You can choose the options for answer incoming calls during office hours.

4）Outside Office Hours

You can choose the options for answer incoming calls during offwork time.

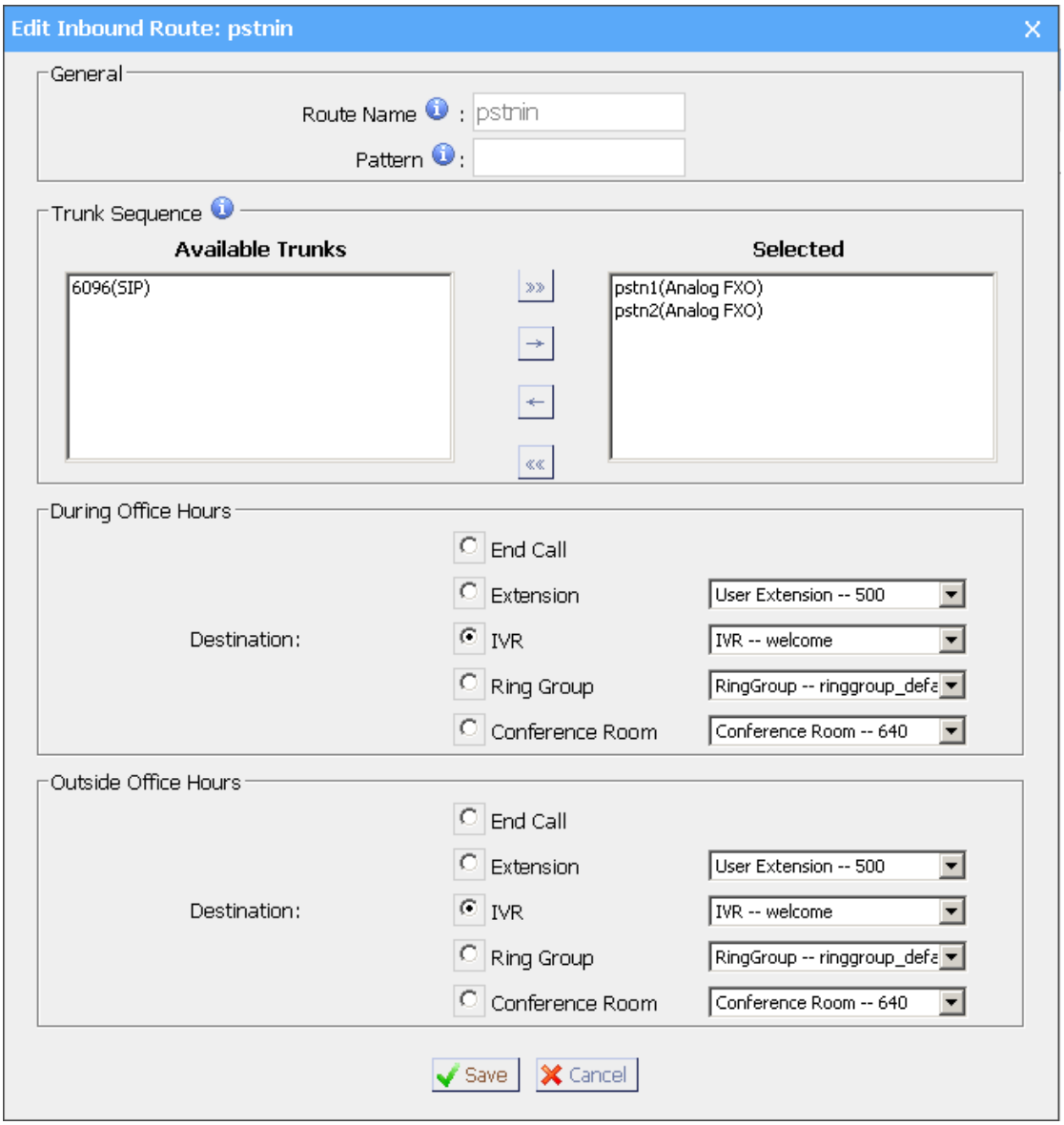


Figure 3-19

3.4.4.2 Delete Routes

Select the item you'd like to remove and click 'Delete' key.

3.5 Other Settings

3.5.1 Conferencing

Clik 'New Conference Room' to enter into page

·Extension

This is the number dialed to reach this Conference Room.

·Pin Code

set an optional pin code, Ex: "somepincode" that must be entered in order to access the Conference Room.

Figure 3-20

3.5.2 Time Conditions

You can setup the business hours here.

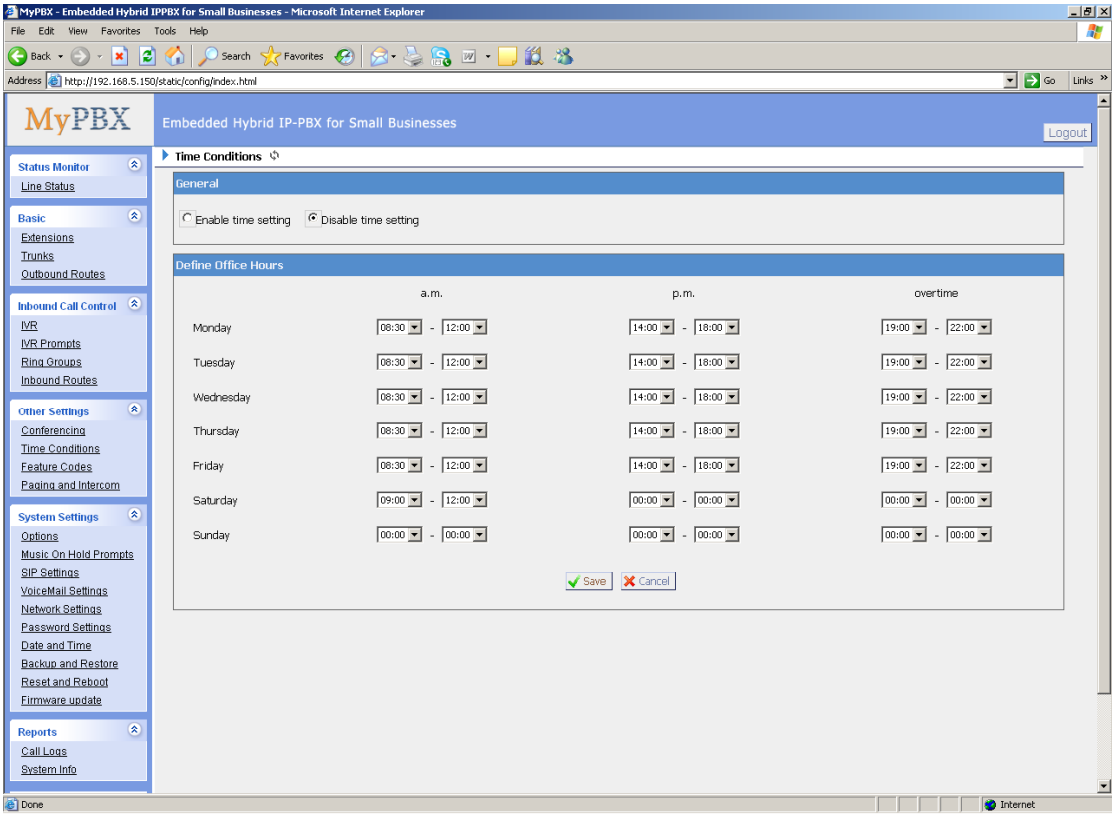


Figure 3-21

3.5.3 Feature Codes

1. General

·One Touch Record

User may do call recording by calling *1 on his phone during a call. (*1 is default setting)

·Extension For Checking Voice Mail

User may checking his voice mail by calling *2 on his phone (*2 is default setting)

·Blind Transfer

User may blind transfer the incoming call by calling this feature code on his phone (disable by default)

·Attended Transfer

User may transfer the incoming call by calling *3 on his phone (*3 is default setting)

·Call Pickup

User may pickup the incoming call by calling *4 on his phone (*4 is default setting)

·Intercom

Define the feature code that is used to prefix an extension to dial it as Intercom. For instance setting this value to *5 would allow to initiate an intercom call with extension 501 by dialing *5501.

3. Call Forward Preferences

·Reset to defaults
User may reset the call forward default value by calling *70 on his phone (*70 is default setting)
Note: When reset to defaults, the call forward default value will be:
Always forward: Disable.
Busy forward to voice mail: Enable
No answer forward to voice mail: Enable
Do not disturb: Disable

·Enable always forward
User may enable always forward by calling *71 on his phone (*71 is default setting)

·Disable always forward
User may disable always forward by calling *071 on his phone (*071 is default setting)

·Enable busy forward
User may enable busy forward by calling *72 on his phone (*72 is default setting)

·Disable busy forward
User may disable busy forward by calling *072 on his phone (*072 is default setting)

·Enable no answer forward
User may enable no answer forward by calling *73 on his phone (*73 is default setting)

·Disable no answer forward
User may disable no answer forward by calling *073 on his phone (*072 is default setting)

·Forward to number
User may forward the call to number by calling *74 on his phone (*74 is default setting)
Note: User may activate call forward to number by dialing *74 + phone number. e.g.: by dialing *74501, it will forward the call to extension 501.

·Forward to Voicemail
User may forward the call to voice mail by calling *074 on his phone (*074 is

default setting)

·Enable do not disturb

User may enable do not disturb by calling *75 on his phone (*75 is default setting)

·Disable do not disturb

User may disable do not disturb by calling *075 on his phone (*075 is default setting)

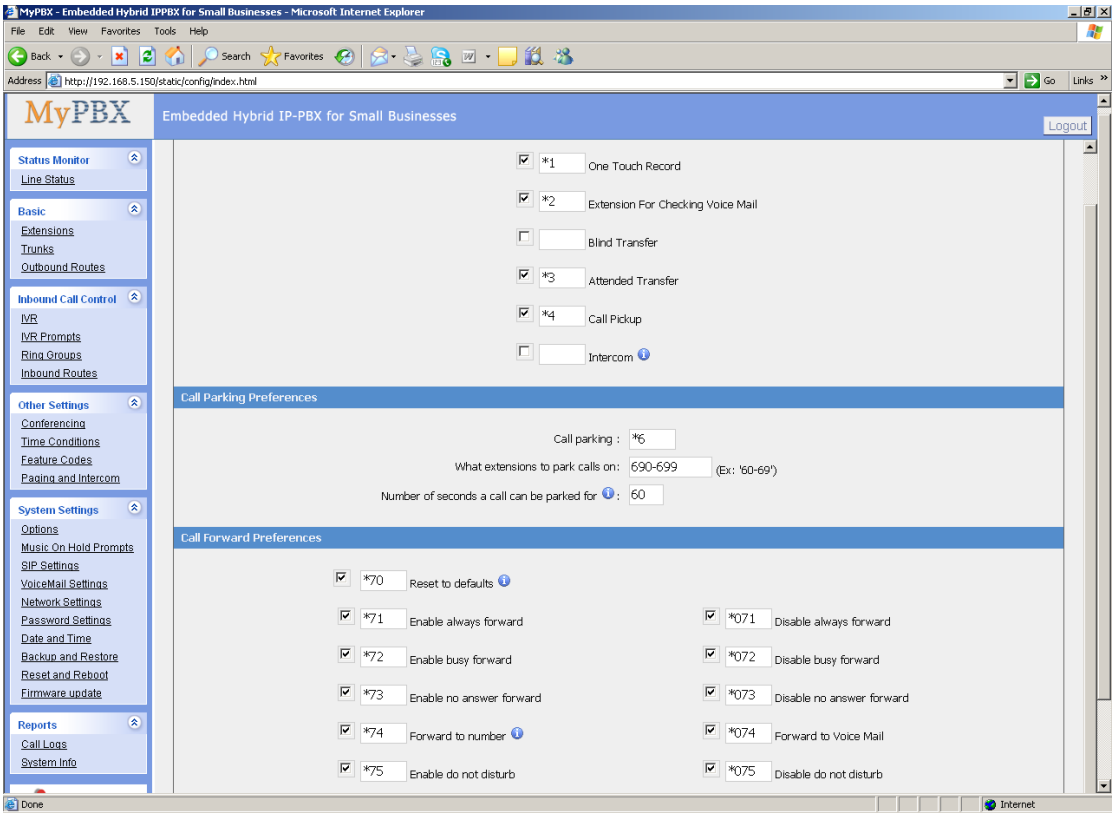


Figure 3-22

3.5.4 Paging and Intercom

Paging and Intercom is for specific phones that are capable of Paging or Intercom. It can be used to make an announcement over the speakerphone on a phone or a group of phones. Targeted phones will not ring, but answer immediately into speaker-phone mode. This section is for configuring group paging, intercom is configured through Feature Codes.

This feature should work with Yealink's T28,T26,T22,T20,T10T,T9CM and possibly other manufacturer's SIP phones(not ATAs). Any phone that is always set to auto-answer should also work.

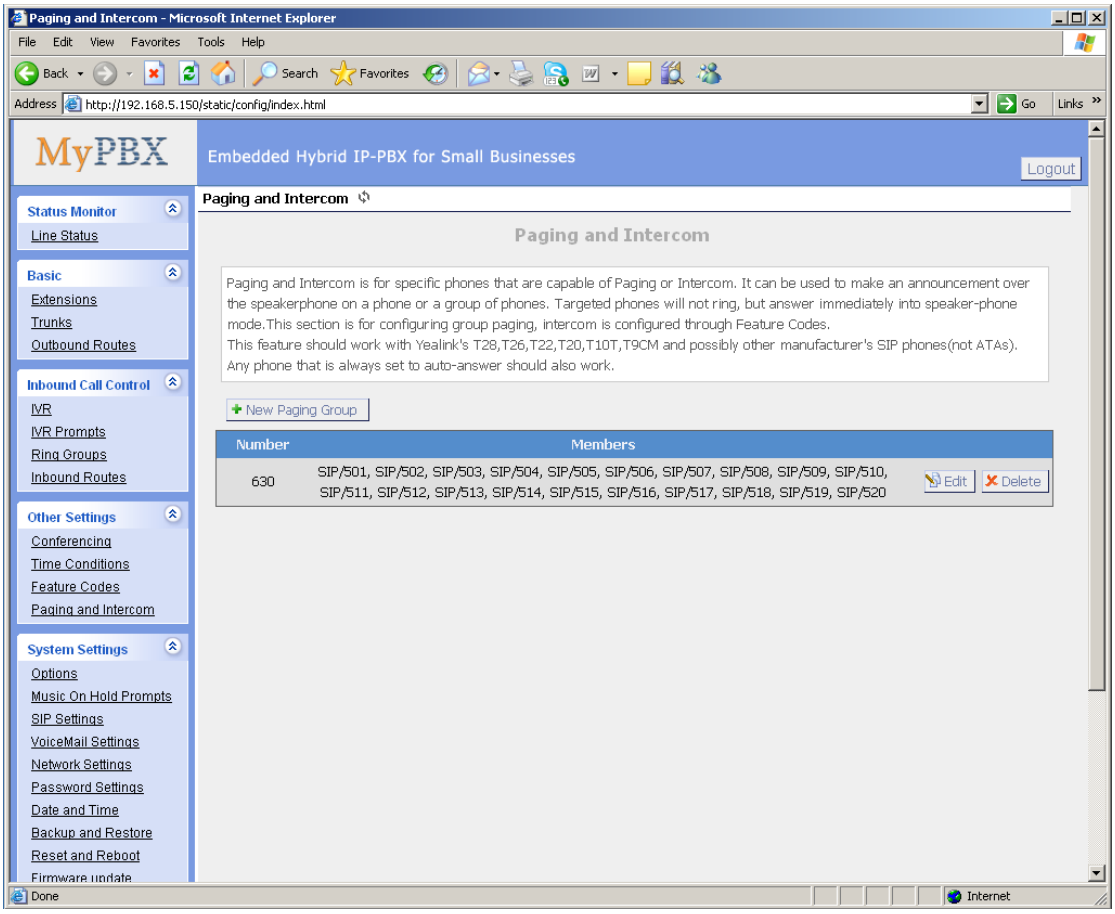

Paging and Intercom - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Address http://192.168.5.150/static/config/index.html
MyPBX
Embedded Hybrid IP-PBX for Small Businesses
Logout
Status Monitor
Line Status
Basic
Extensions
Trunks
Outbound Routes
Inbound Call Control
IVR
IVR Prompts
Ring Groups
Inbound Routes
Other Settings
Conferencing
Time Conditions
Feature Codes
Paging and Intercom
System Settings
Options
Music On Hold Prompts
SIP Settings
VoiceMail Settings
Network Settings
Password Settings
Date and Time
Backup and Restore
Reset and Reboot
Firmware update
Paging and Intercom
Paging and Intercom
Paging and Intercom is for specific phones that are capable of Paging or Intercom. It can be used to make an announcement over the speakerphone on a phone or a group of phones. Targeted phones will not ring, but answer immediately into speaker-phone mode.This section is for configuring group paging, intercom is configured through Feature Codes.
This feature should work with Yealink's T28,T26,T22,T20,T10T,T9CM and possibly other manufacturer's SIP phones(not ATAs).
Any phone that is always set to auto-answer should also work.
New Paging Group
Number
Members
630
SIP/501, SIP/502, SIP/503, SIP/504, SIP/505, SIP/506, SIP/507, SIP/508, SIP/509, SIP/510, SIP/511, SIP/512, SIP/513, SIP/514, SIP/515, SIP/516, SIP/517, SIP/518, SIP/519, SIP/520
Edit
Delete
Done
Internet

Figure 3-23

Click 'New Paging Group' to enter into page.

·Number for this Paging group

This option defines the numbered extension that may be dialed to reach this paging Group.

·Duplex

Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it like an "instant conference".

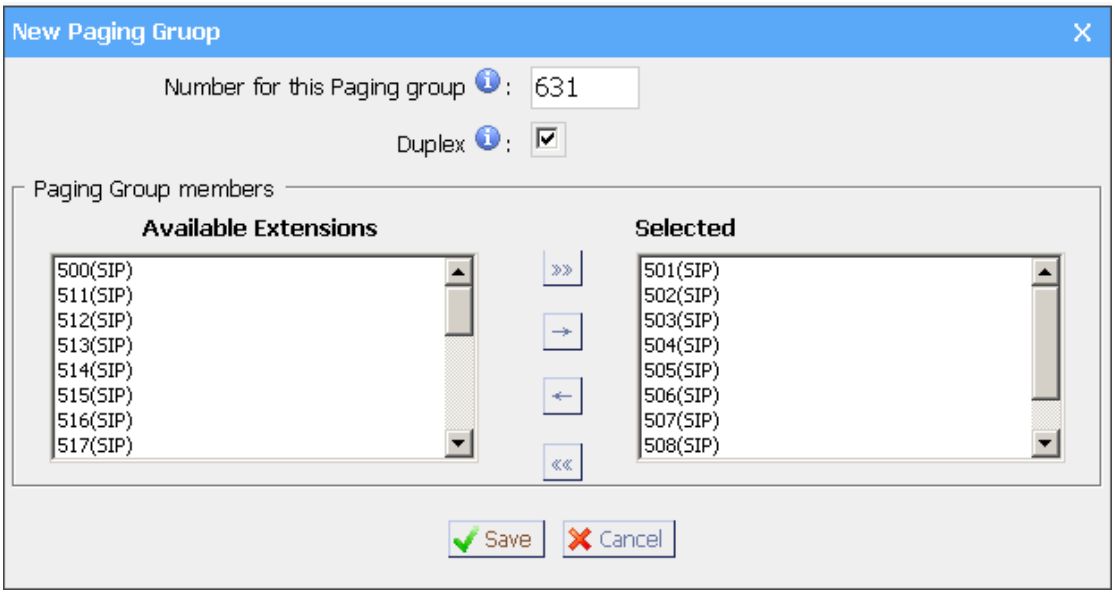


Figure 3-24

3.6 System Settings

3.6.1 Options

1) General

·Ring Timeout

Number of seconds to ring a device before sending to user's Voicemail Box.

·Music on hold

User can select wanted music for holding a call.

·Tone Region

Please choose your country or your nearest neighboring country for default Tones (Ex: dialtone, busy tone, ring tone etc.)

2) Extension Preferences

·User Extensions

The default value is 500 to 616

·Paging Group Extensions

The default value is 620 to 629

·Ring Group Extensions

The default value is 630 to 639

·Conference Extensions

The default value is 640 to 659

·IVR Extensions

The default value is 660 to 689

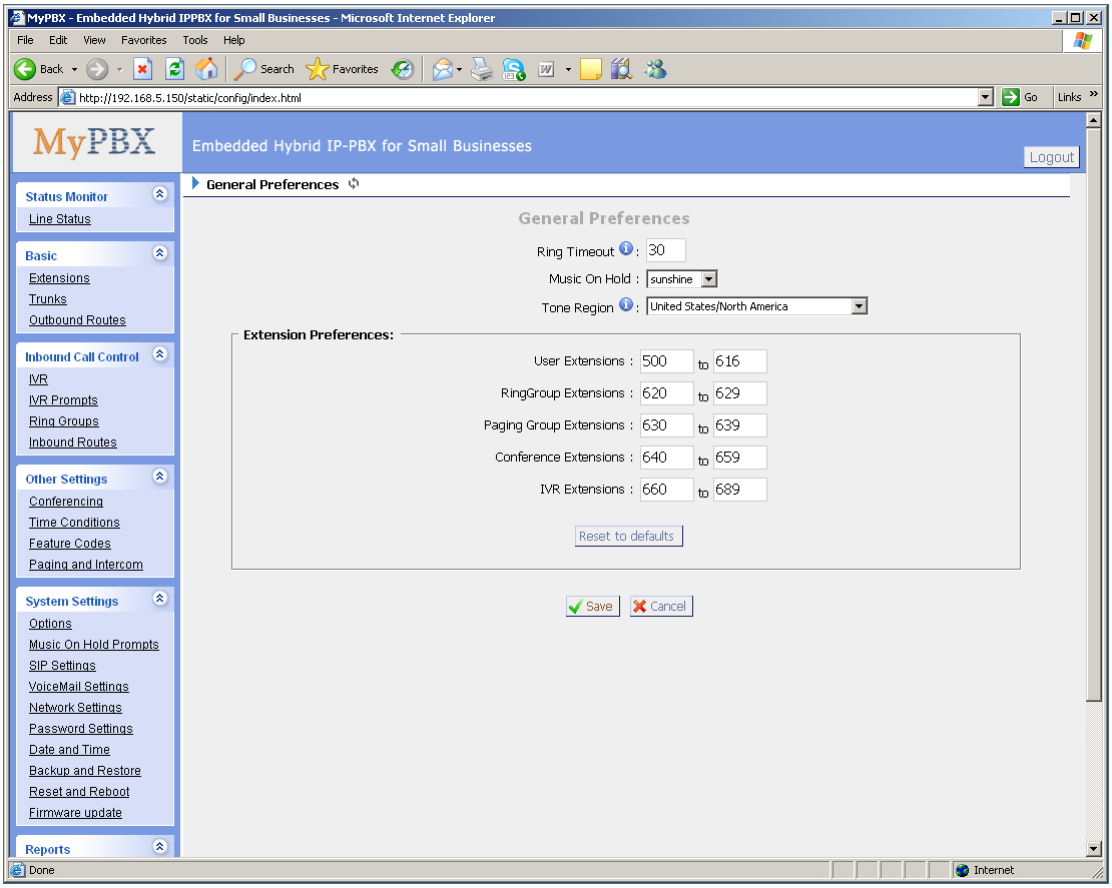


Figure 3-25

3.6.2 Music on Hold Prompts

Admin can upload on hold music as follow:

1) Click 'Upload Music On hold Prompt'.

2) Click 'Browse' to choose the on hold music Prompt.

3) Click 'Upload' to upload the selected Prompt.

Note: The sound file format: GSM 6.10, 8.000kHz, Mono, 1kb/sec

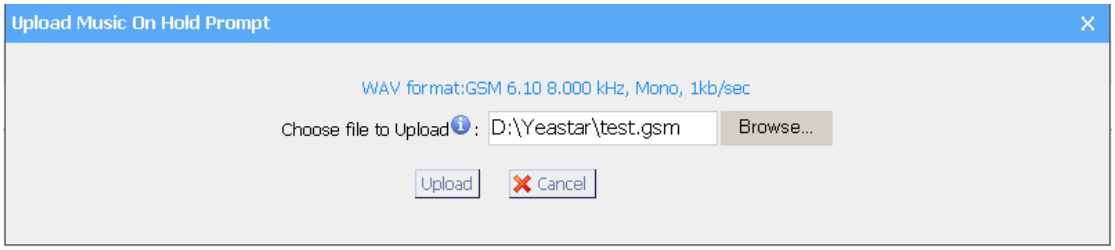


Figure 3-26

3.6.3 SIP Settings

1) General

·UDP Port

UDP Port to bind to, SIP standard port is 5060.

·RTP Port Start

RTP Port start with

·RTP Port End

RTP Port end with

·DTMF Mode

Set default dtmf mode for sending DTMF, default: rfc2833

·Max Registration/Subscription Time

Maximum duration (in seconds) of incoming registration/subscriptions we allow. Default 3600 seconds.

·Min Registration/Subscription Time

Minimum duration (in seconds) of registrations/subscriptions. Default 60 seconds

·Default Incoming/Outgoing Registration Time

Default duration (in seconds) of incoming/outoing registration

2) NAT

·Extern IP

Address that we're going to put in outbound SIP messages if we're behind a NAT

·Extern Host

Alternatively you can specify an external host, and MyPBX will perform DNS queries periodically. Not recommended for production environments! Use external instead.

·Extern Refresh

How often to refresh extern host if used. You may specify a local network in the field below

·Local Network Address

'192.168.0.0/255.255.0.0' : All RFC 1918 addresses are local networks;
'10.0.0.0/255.0.0.0' : Also RFC1918;
'172.16.0.0/12':Another RFC1918 with CIDR notation;
'169.254.0.0/255.255.0.0' : Zero conf local network.

·NAT Mode

Global NAT settings (Affects all peers and users);
yes = Always ignore info and assume NAT;
no = Use NAT mode only according to RFC3581;
never = Never attempt NAT mode or RFC3581 support;
route = Assume NAT, don't send rport

·Allow RTP Reinvite

MyPBX by default tries to redirect the RTP media stream (audio) to go directly from the caller to the callee. Some devices do not support this (especially if one of them is behind a NAT).

3）Codecs

A codec is a compression or decompression algorithm run against voice as it is moved between analog (speaking) and digital (VoIP).

u-law: A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64kbit/s for each direction (receiving and transmitting) of a VoIP call. u-law should be supported by all VoIP phones.

a-law: A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s for each direction (receiving and transmitting) of a VoIP call. a-law should be supported by all VoIP phones.

GSM: A wireless standard codec, used worldwide, that provides okay voice quality and consumes 13.3kbit/s for each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

G.726: A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s for each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

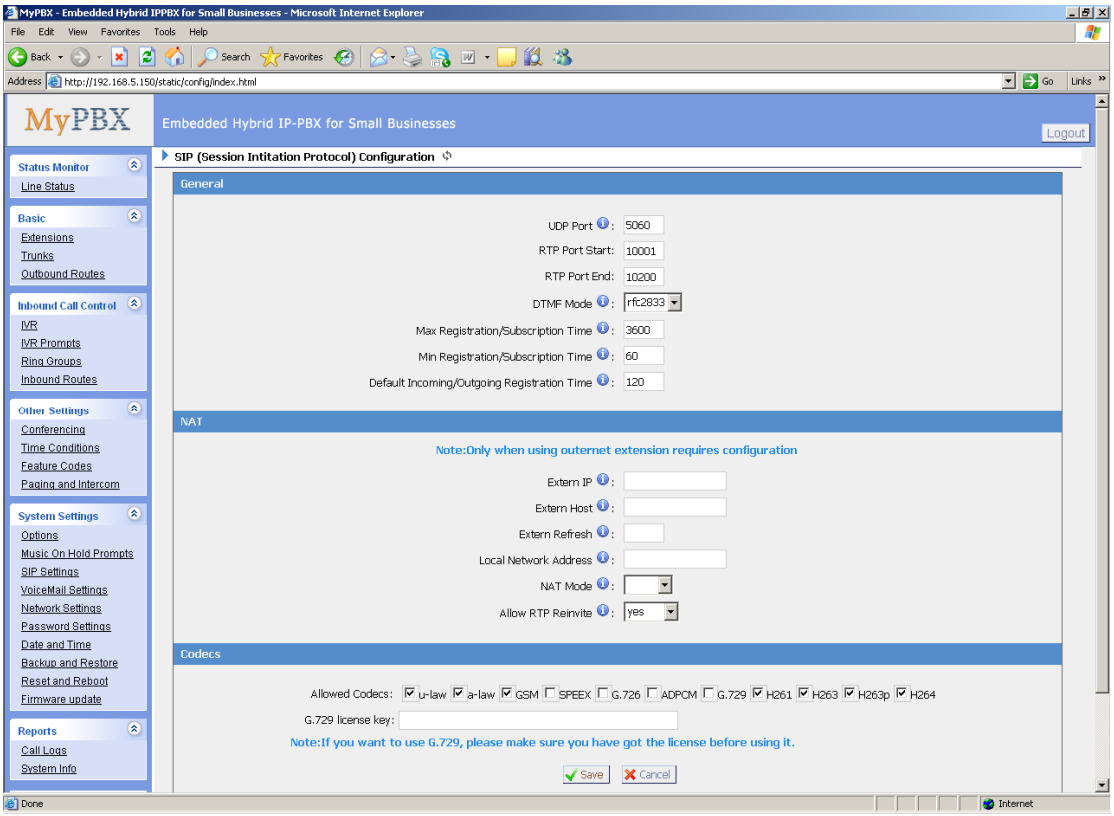


Figure 3-27

3.6.4 Voicemail Settings

General Voicemail Settings

·Maximum messages per folder

This select box sets the maximum number of messages that a user may have in any of their folders.

·Max message time

This select box sets the maximum duration of a voicemail message in minutes. Message recording will not occur for times greater than this amount.

·Min message time

This select box sets the minimum duration of a voicemail message in seconds. Messages below this threshold will be automatically deleted.

·Say message Caller-ID

If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.

·Say message duration

If this option is set, the duration of the message in minutes will be played back before the voicemail message begins playing.

.Say message arrival time

If this option is set, the arrival time of the message will be played back before the voicemail message begins playing.

SMTP Settings for Voicemail

·E-mail Address

The E-mail Address that MyPBX will use to send voicemail to users' email.

·Password

The E-mail password.

·SMTP Server

The IP address or hostname of an SMTP server that the MyPBX may connect to, in order to send e-mail notifications of your voicemails;
i.e. mail.yourcompany.com

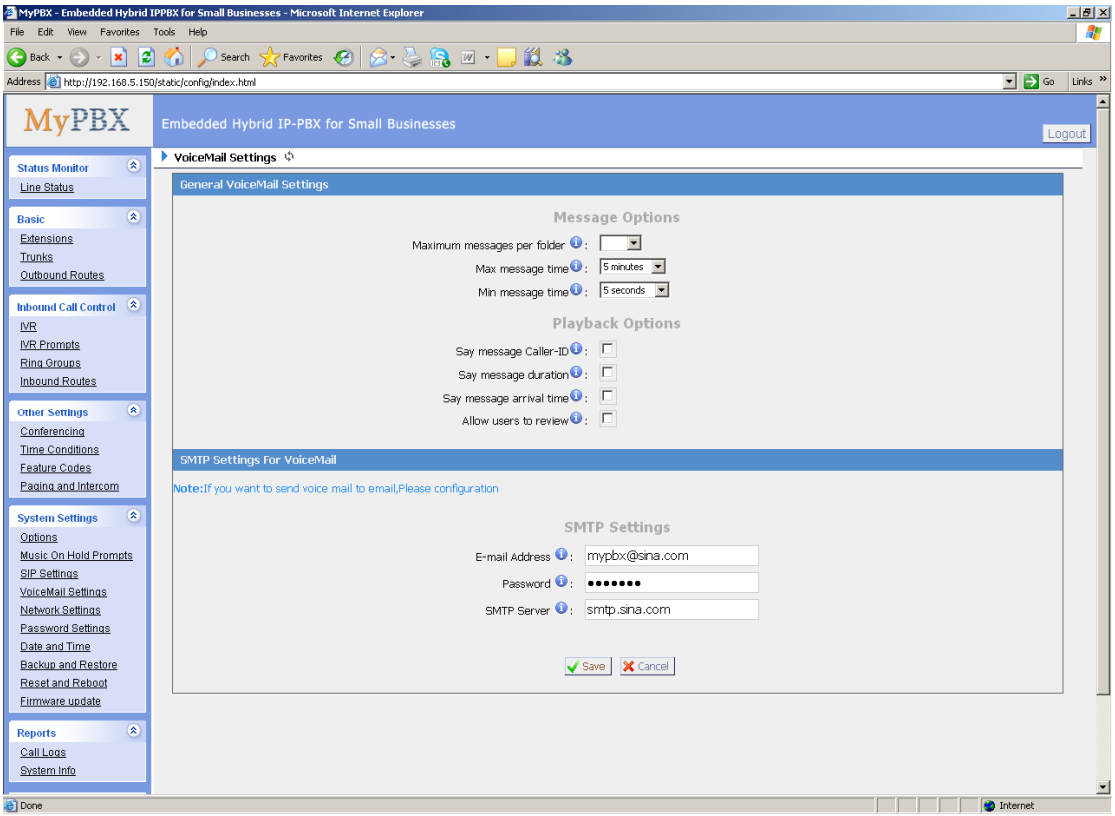


Figure 3-28

3.6.5 Network Setting

·DHCP

If this option is set, MyPBX will use DHCP to get an available IP address from your local network.

·Hostname

Set the host name for MyPBX Appliance.

·IP Address

Set the IP Address for MyPBX Appliance.

·Subnet Mask

Set the Subnet Mask for MyPBX Appliance.

·Gateway

Set the Gateway for MyPBX Appliance.

·Primary DNS

Set the primary DNS for MyPBX Appliance.

·Secondary DNS

Set the secondary DNS for MyPBX Appliance.

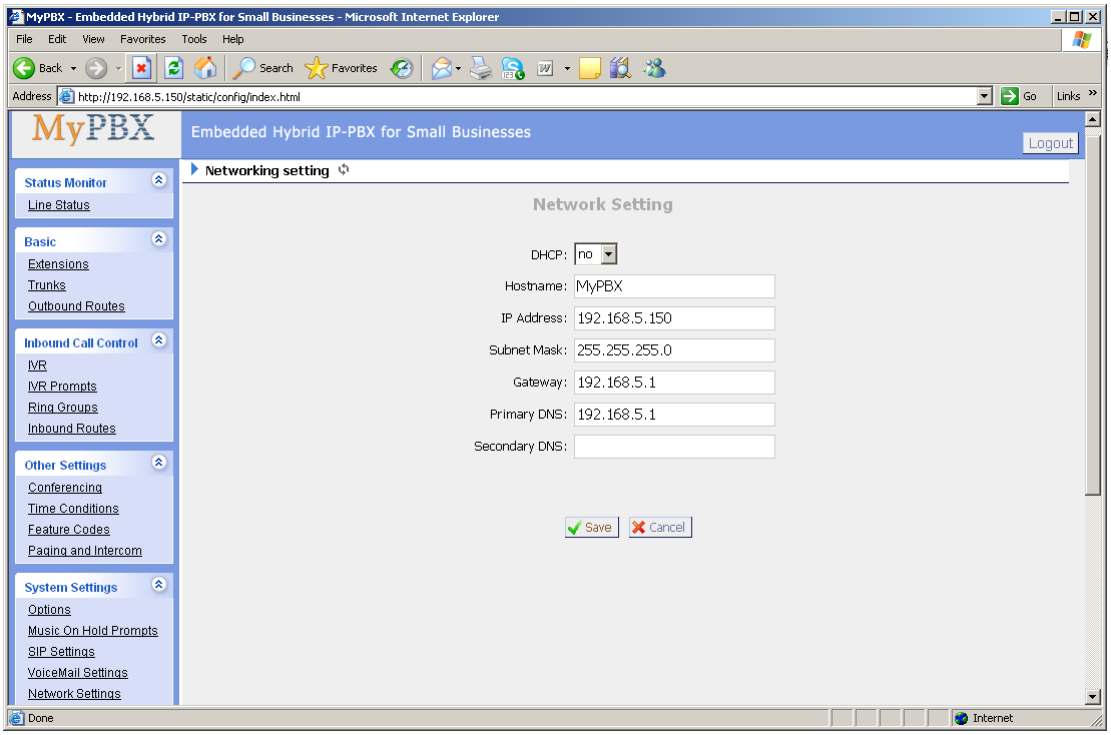


Figure 3-29

3.6.6 Password Settings

The default password is '**password**', we need change that first.
Enter new password and click update, then the system will prompt you re-login using your new password.

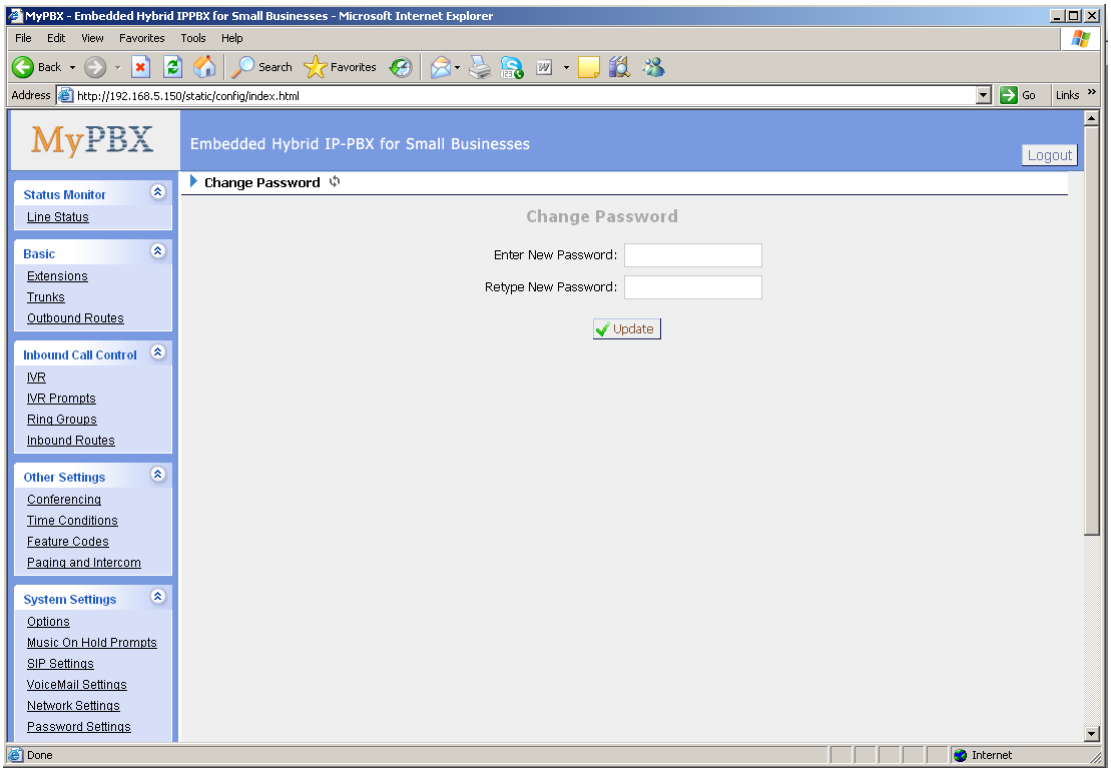


Figure 3-30

3.6.7 Date and Time

Set the date and time for MyPBX Appliance.

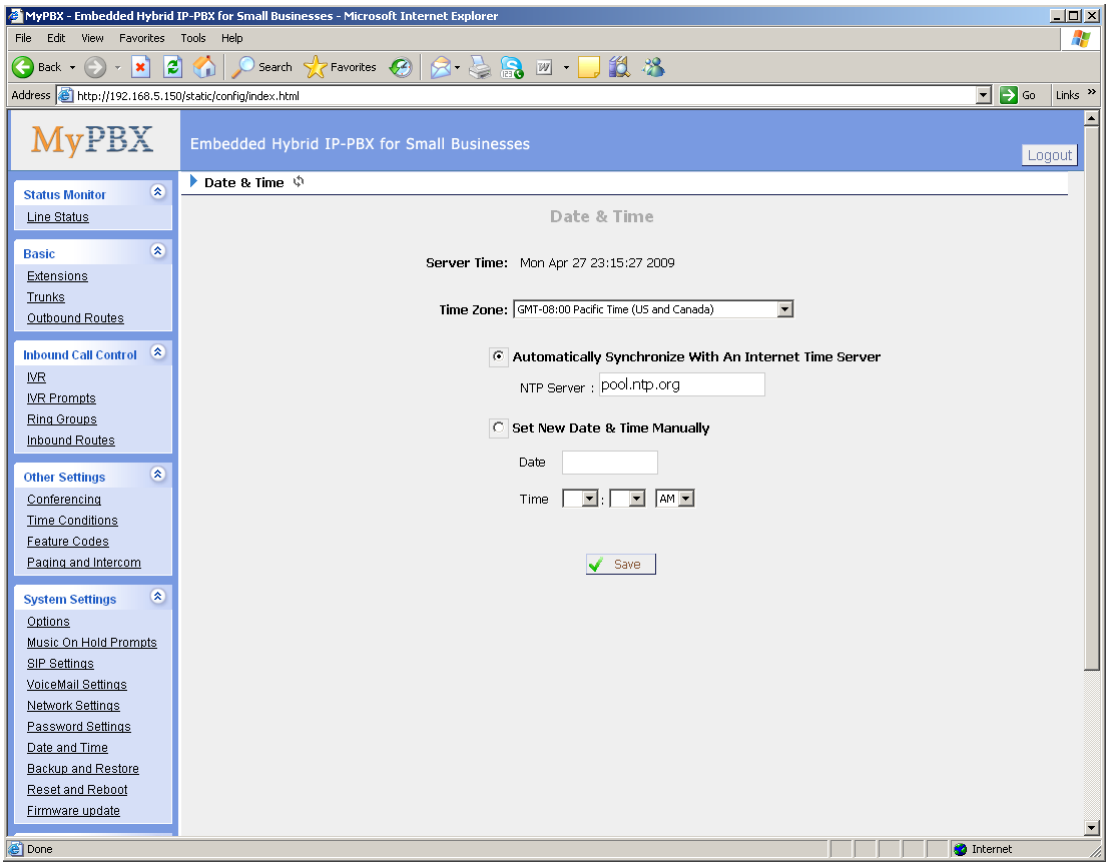


Figure 3-31

3.6.8 Backup and Restore

Backup / Restore MyPBX's Configuration.

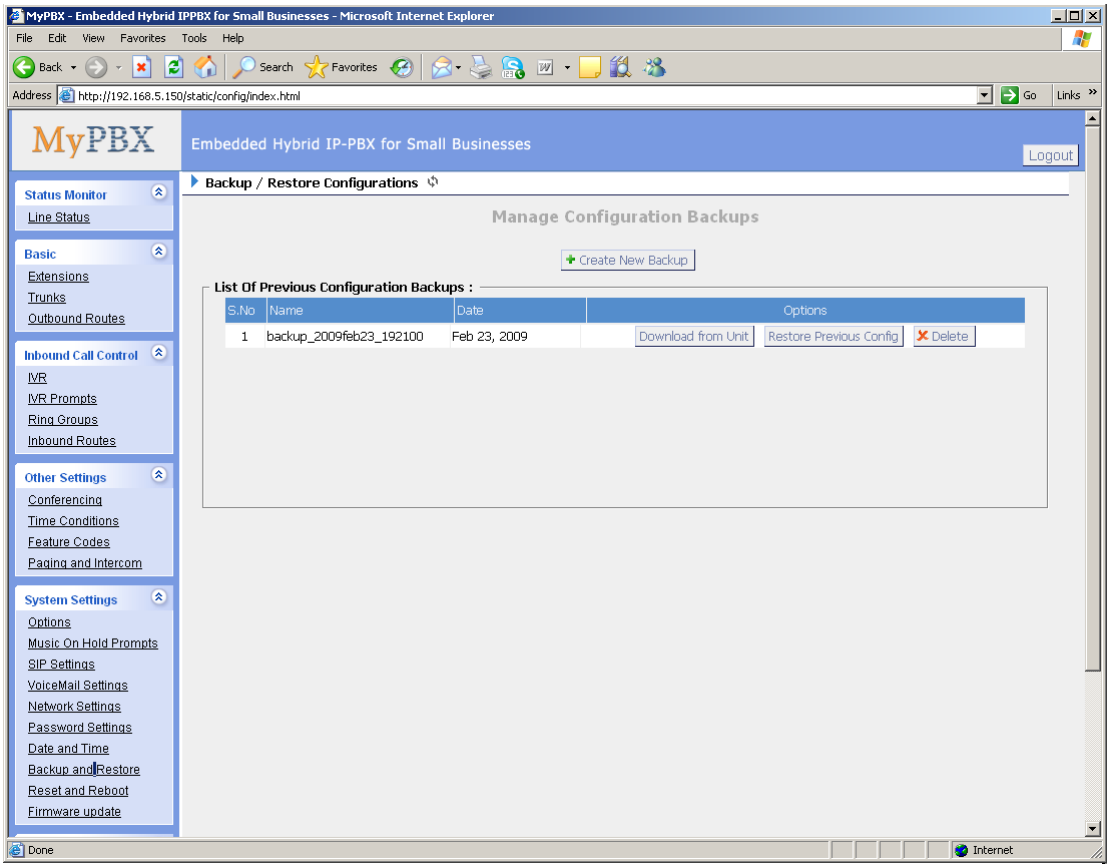


Figure 3-32

3.6.9 Reset and Reboot

·Reboot Appliance

Warning: Rebooting the appliance will terminate all active calls.

·Reset to Factory Defaults

Warning: By resetting your System to factory defaults, you will lose all your configurations! You can't cut off the electricity during restarting till the RUN state light blink, otherwise it will cause the failure performance.

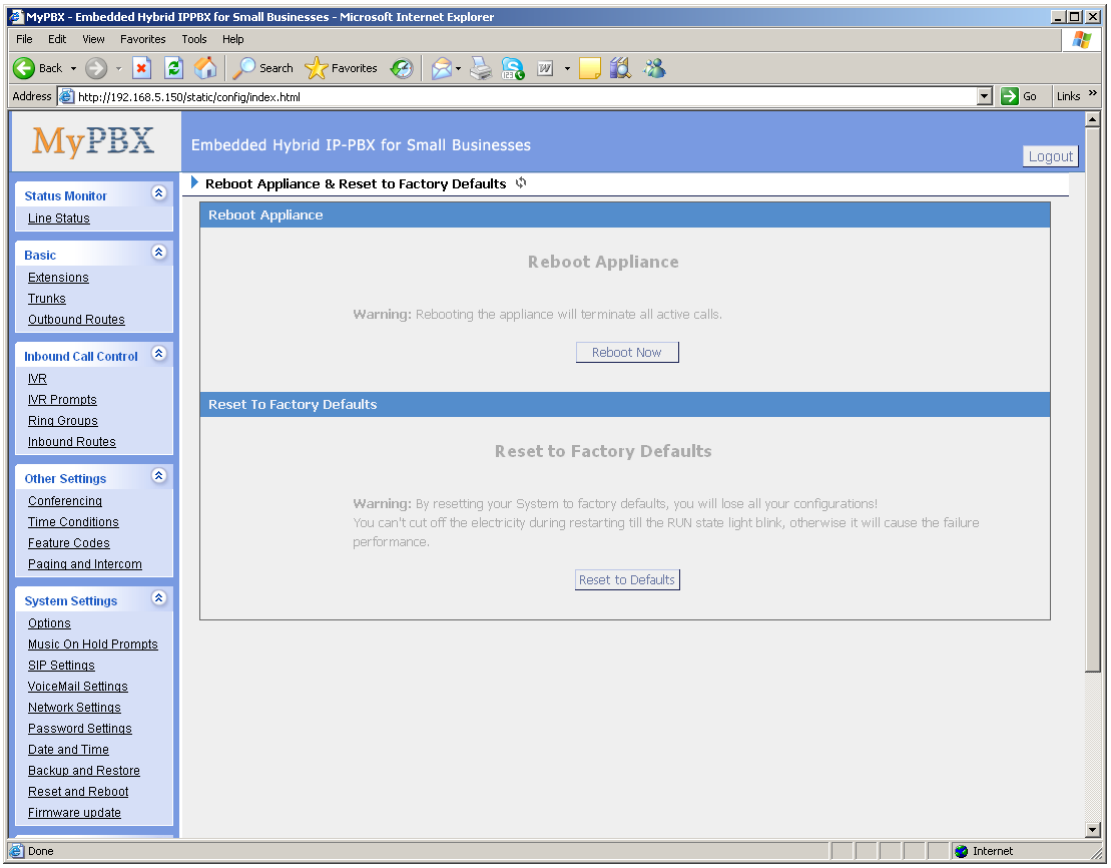


Figure 3-33

3.6.10 Firmware Update

With new version, we can upgrade the firmware in GUI through a TFTP Server or HTTP URL.
Enter your TFTP Server ip address and firmware name, just click 'start', you will update your firmware easily.

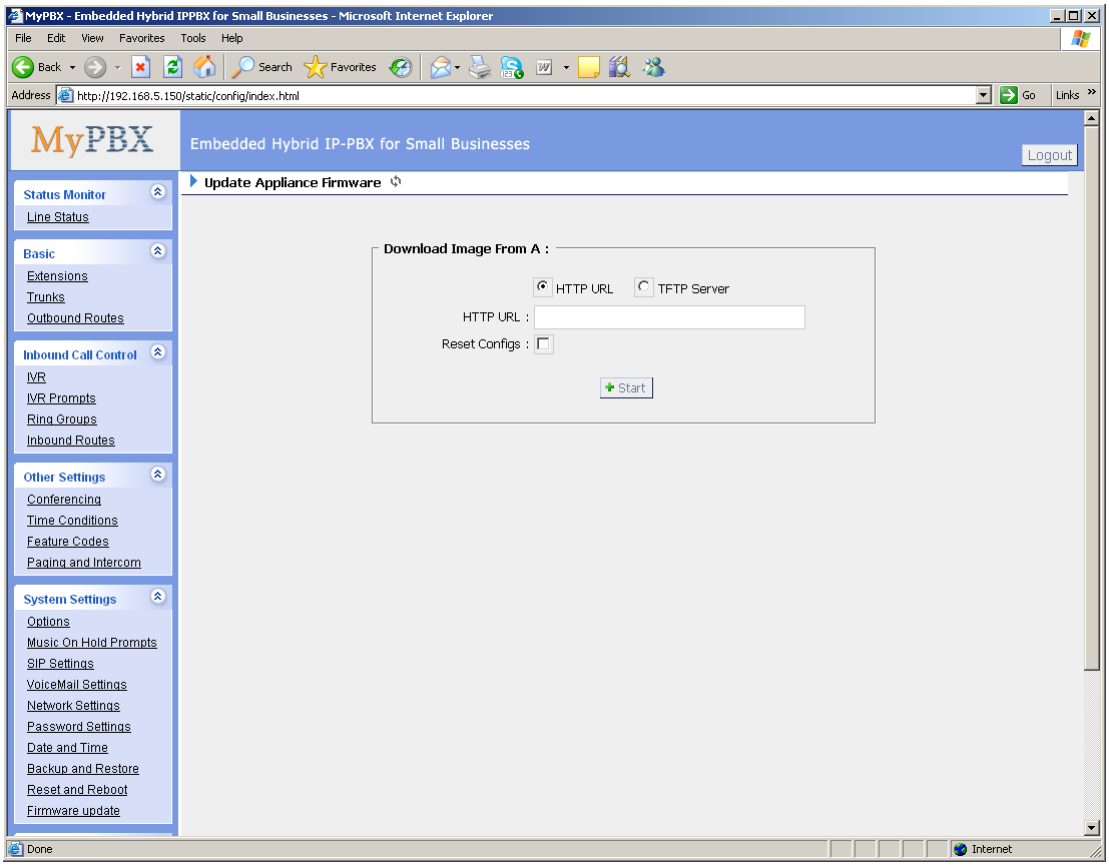


Figure 3-34

3.7 Reports

3.7.1 Call Logs

Call Log captures all call details: calling time, caller number, callee number, call type, call duration and so on. For administrator's convenience, admin can search information by calling time, call type and call duration and export them into CSV file.

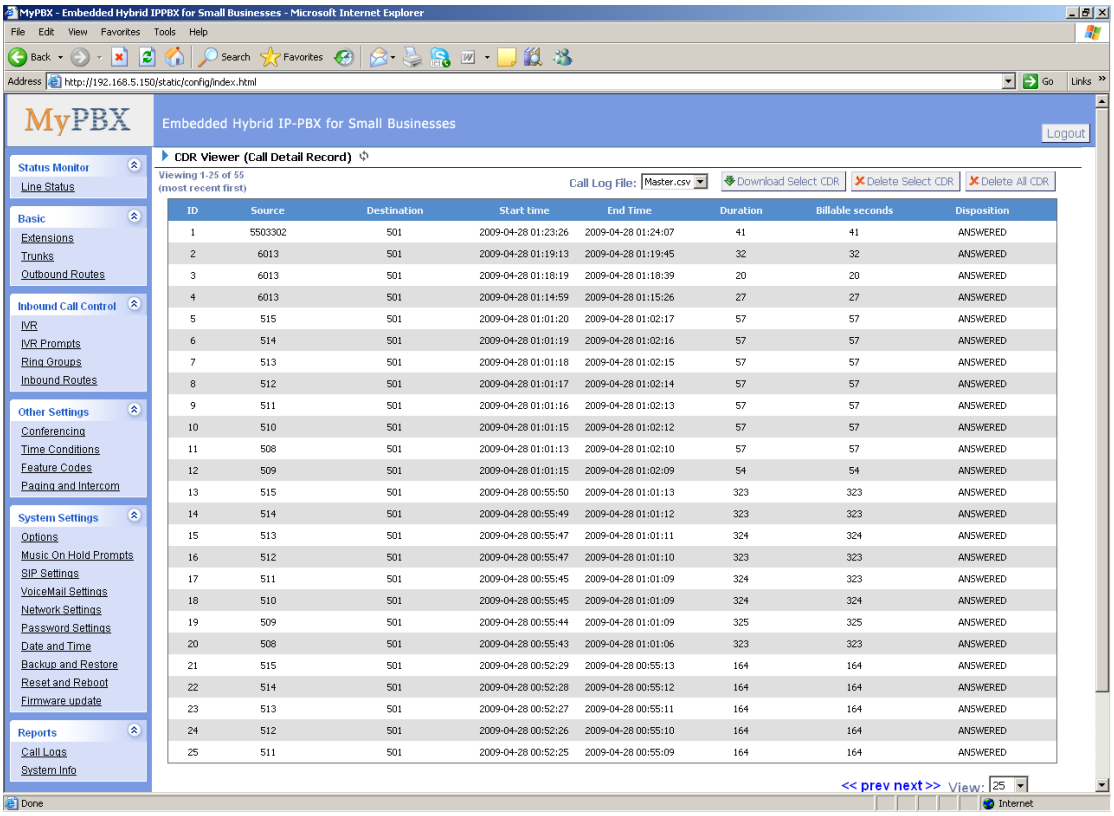


Figure 3-35

3.7.2 System Info

General:
Information about Hardware version, Firmware version and Uptime.

Network:
Information about Hostname, MAC, IP Address, Mask, Gateway and DNS.

Disk Usage:
Disk usage information.

Memory Usage:
Memory usage information.

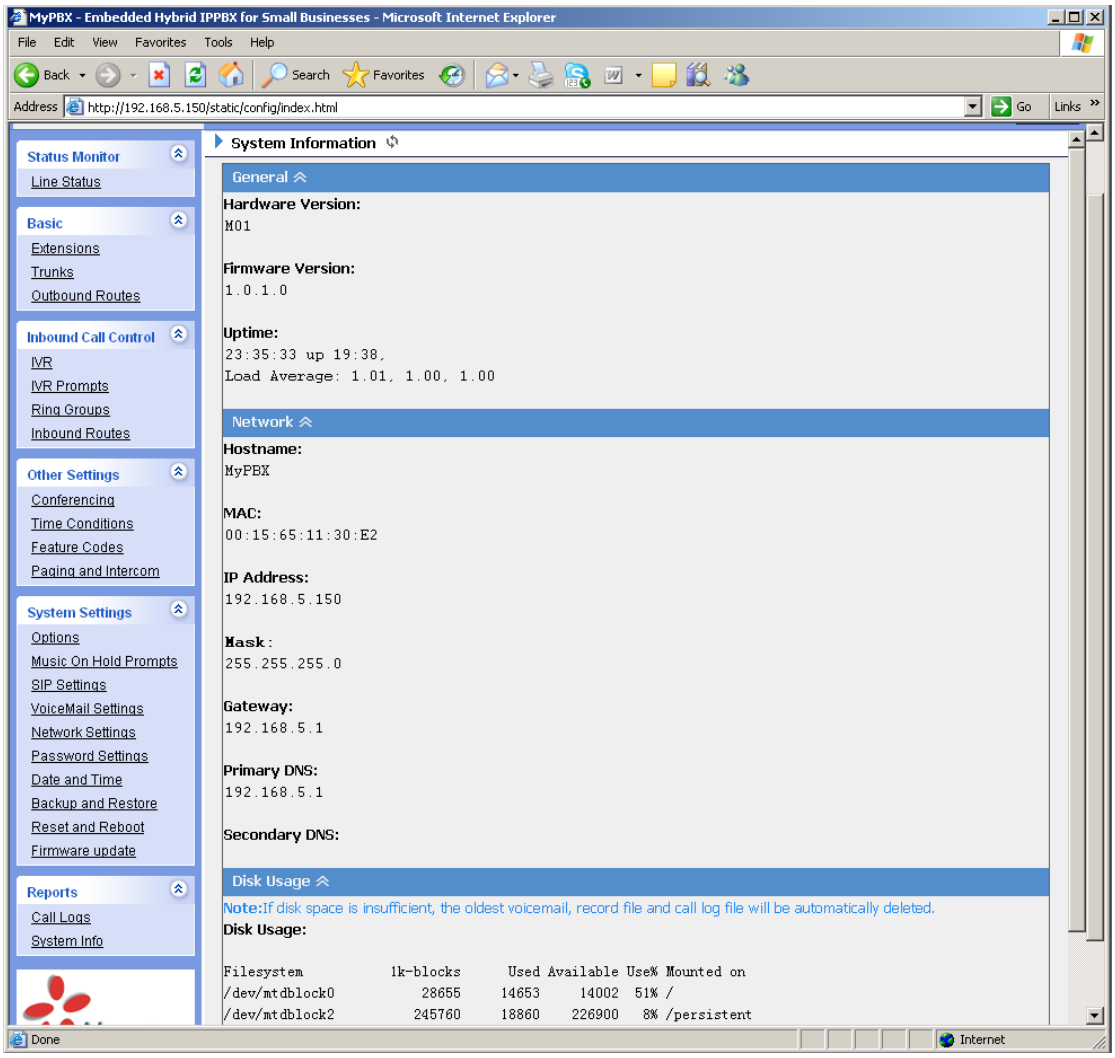

MyPBX - Embedded Hybrid IPPBX for Small Businesses - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://192.168.5.150/static/config/index.html
Go Links
Status Monitor
Line Status
Basic
Extensions
Trunks
Outbound Routes
Inbound Call Control
IVR
IVR Prompts
Ring Groups
Inbound Routes
Other Settings
Conferencing
Time Conditions
Feature Codes
Paging and Intercom
System Settings
Options
Music On Hold Prompts
SIP Settings
VoiceMail Settings
Network Settings
Password Settings
Date and Time
Backup and Restore
Reset and Reboot
Firmware update
Reports
Call Logs
System Info
System Information
General
Hardware Version:
M01
Firmware Version:
1.0.1.0
Uptime:
23:35:33 up 19:38,
Load Average: 1.01, 1.00, 1.00
Network
Hostname:
MyPBX
MAC:
00:15:65:11:30:E2
IP Address:
192.168.5.150
Mask:
255.255.255.0
Gateway:
192.168.5.1
Primary DNS:
192.168.5.1
Secondary DNS:
Disk Usage
Note:If disk space is insufficient, the oldest voicemail, record file and call log file will be automatically deleted.
Disk Usage:
Filesystem 1k-blocks Used Available Use% Mounted on
/dev/mtdblock0 28655 14653 14002 51% /
/dev/mtdblock2 245760 18860 226900 8% /persistent
Done
Internet

Figure 3-36

4 Access MRI

MRI(MyPBX Recording Interface: Check Voice Mail).Users may access MRI by using extension number as username and voicemail secret as password.

4.1 Allow users to access MRI

Extension's option hasmanager must be set to yes before it can access to MRI.

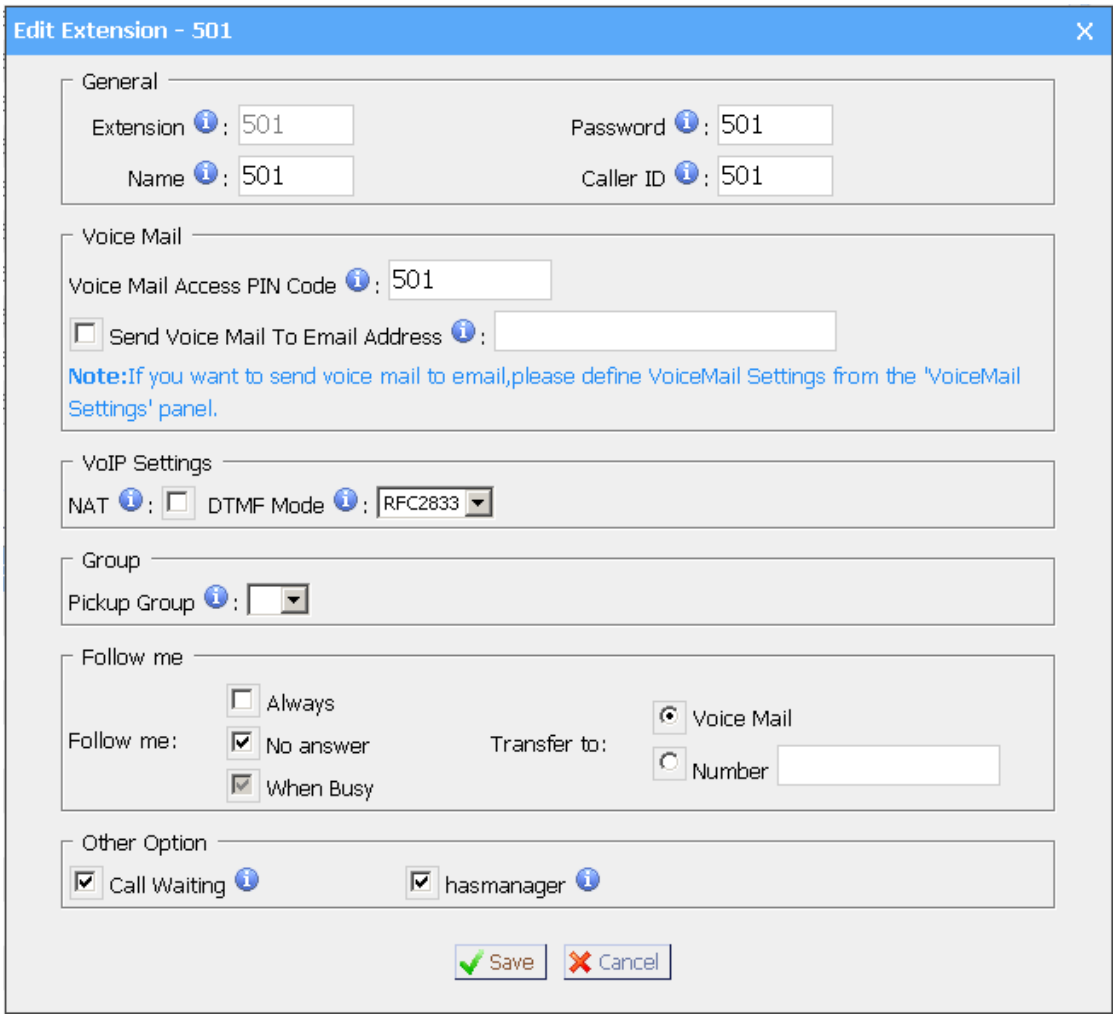


Figure 4-1

4.2 User login

Users can use extension number as MRI username, voicemail password as MRI password to access MRI.

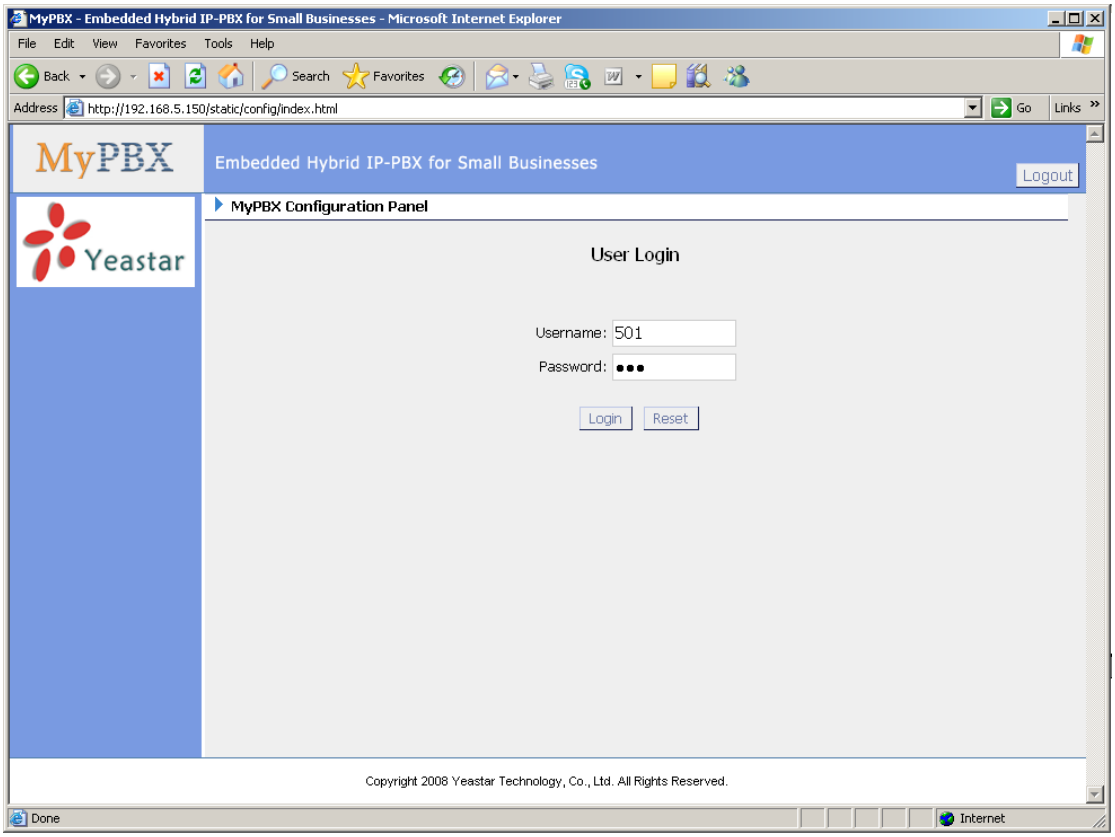


Figure 4-2

4.3 Voicemail

Users can check, delete, move and download voicemail here

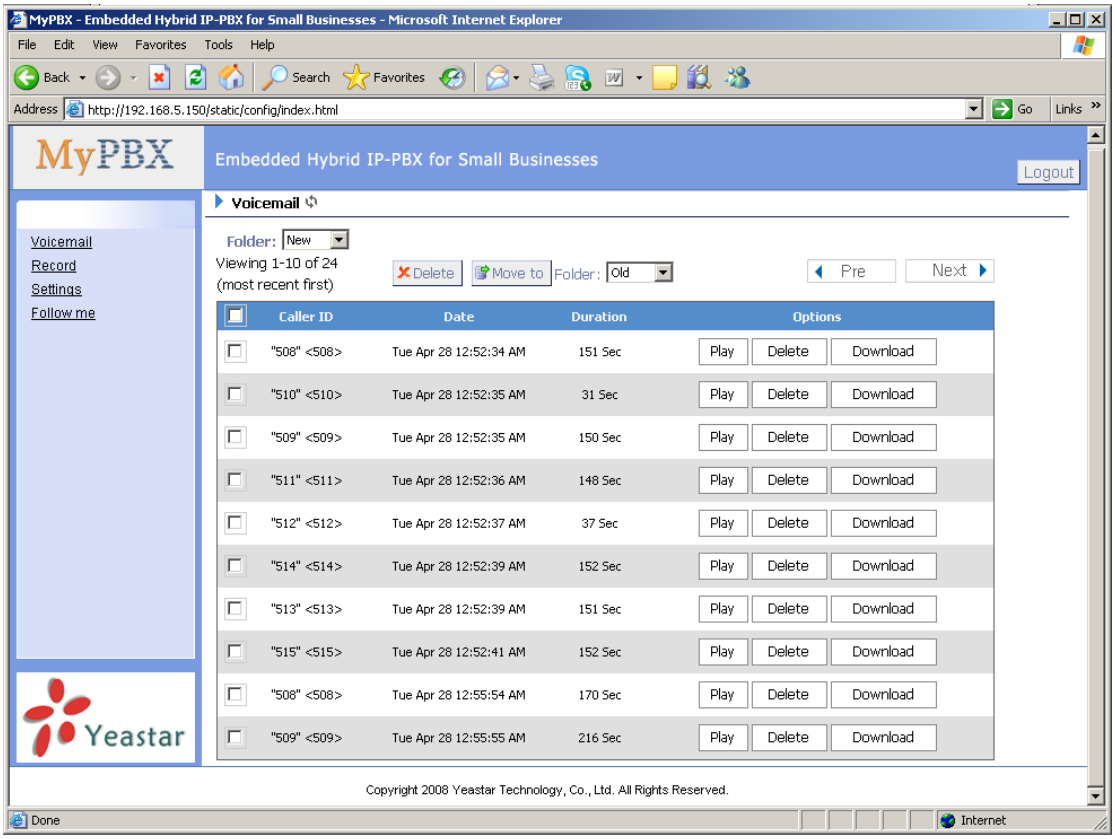

MyPBX - Embedded Hybrid IP-PBX for Small Businesses - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://192.168.5.150/static/config/index.html
Go Links
MyPBX
Embedded Hybrid IP-PBX for Small Businesses
Logout
Voicemail
Record
Settings
Follow me
Voicemail
Folder: New
Viewing 1-10 of 24
(most recent first)
Delete
Move to Folder: Old
Pre
Next
Caller ID
Date
Duration
Options
"508" <508> Tue Apr 28 12:52:34 AM 151 Sec Play Delete Download
"510" <510> Tue Apr 28 12:52:35 AM 31 Sec Play Delete Download
"509" <509> Tue Apr 28 12:52:35 AM 150 Sec Play Delete Download
"511" <511> Tue Apr 28 12:52:36 AM 148 Sec Play Delete Download
"512" <512> Tue Apr 28 12:52:37 AM 37 Sec Play Delete Download
"514" <514> Tue Apr 28 12:52:39 AM 152 Sec Play Delete Download
"513" <513> Tue Apr 28 12:52:39 AM 151 Sec Play Delete Download
"515" <515> Tue Apr 28 12:52:41 AM 152 Sec Play Delete Download
"508" <508> Tue Apr 28 12:55:54 AM 170 Sec Play Delete Download
"509" <509> Tue Apr 28 12:55:55 AM 216 Sec Play Delete Download
Yeastar
Copyright 2008 Yeastar Technology, Co., Ltd. All Rights Reserved.
Done
Internet

Figure 4-3

4.4 Record

Users can play, delete and download record files here

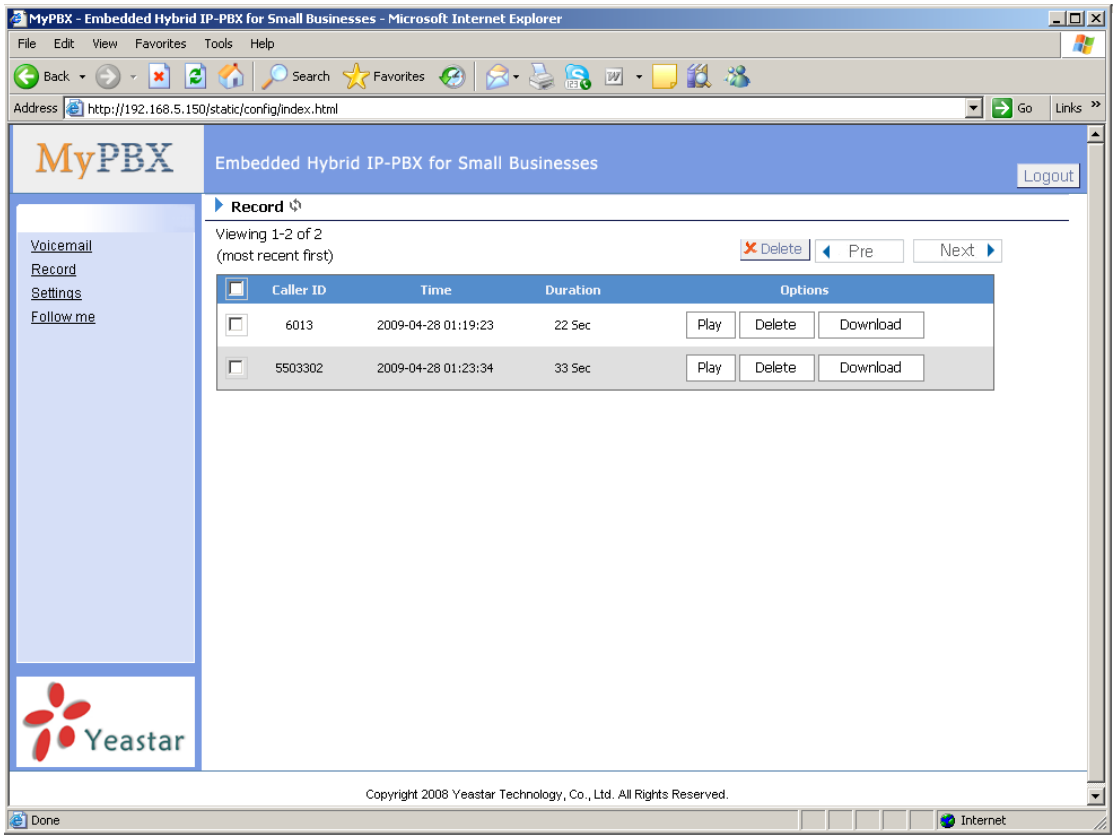


Figure 4-4

4.5 settings

Voicemail password: new voicemail box password.
Enter again to confirm: confirm new voicemail box password.
Send Voice Mail To Email Address: Configure the email address voicemails will be sent to.

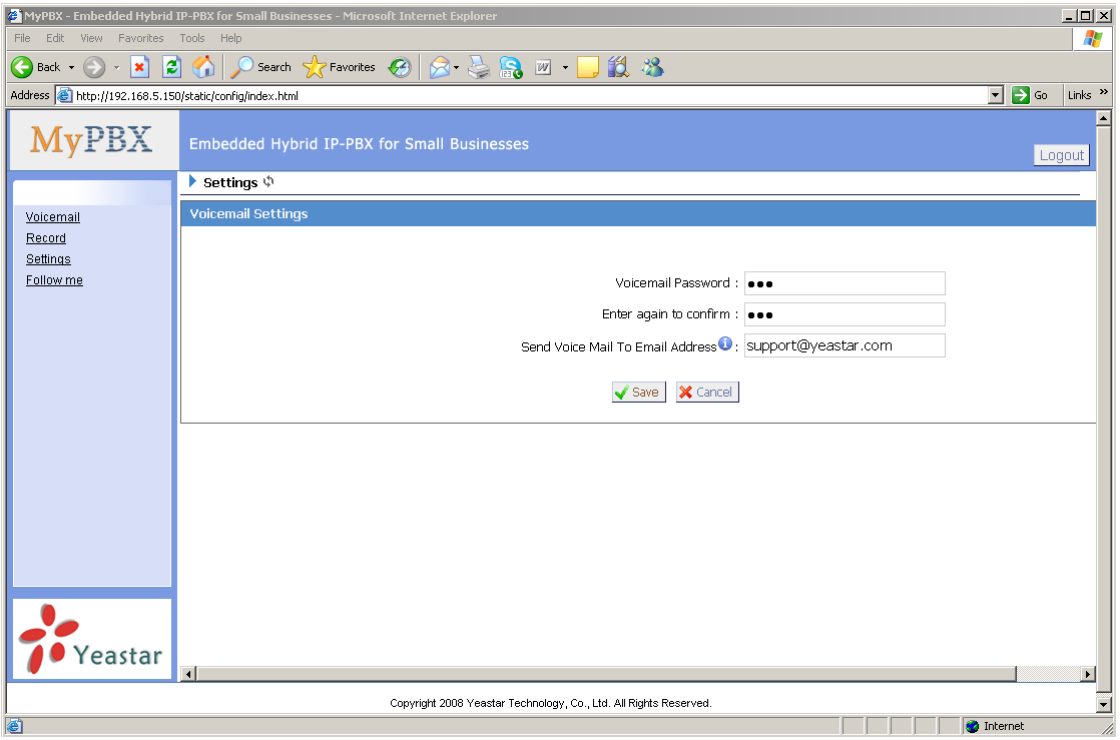
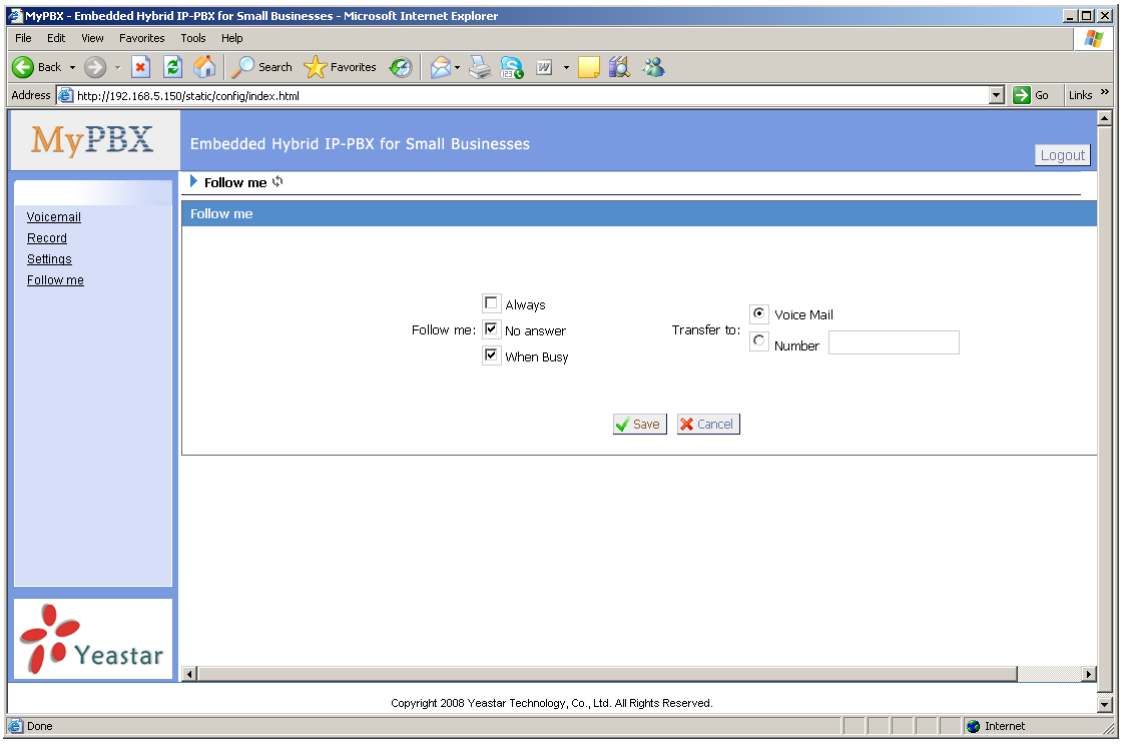


Figure 4-5

4.6 Follow me

Set the mode of call forwarding when there's incoming call on this extension.


MyPBX - Embedded Hybrid IP-PBX for Small Businesses - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Back Search Favorites
Address http://192.168.5.150/static/config/index.html
Go Links
MyPBX
Embedded Hybrid IP-PBX for Small Businesses
Logout
Follow me
Voicemail
Record
Settings
Follow me
Follow me
Always
Follow me: No answer
When Busy
Transfer to: Voice Mail
Number
Save
Cancel
Yeastar
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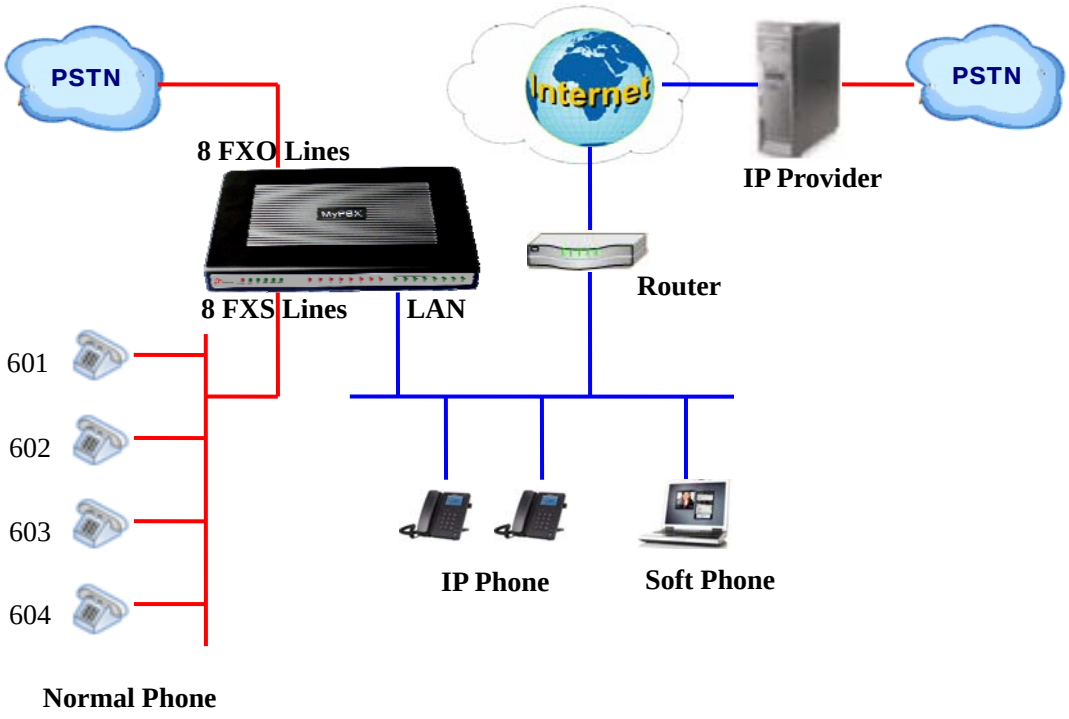
Done
Internet

Figure 4-6

5 Use MyPBX



5.1 Make outbound call

To make an outbound call, we need to add trunk/service provider first. There are two types of service provider:

·**Analog Ports:** FXO ports of MyPBX, connect to local PSTN

·**SIP Trunk:** Connect to remote SIP VOIP service provider server

What are FXO and FXS?

FXS (Foreign eXchange Station) is an interface wihich drives a telephone or FAX machine. FXS interfaces get phones plugged into them, delivery battery, and provide ringing, FXS interfaces are signaled with FXO signaling.

FXO (Foreign eXchange Office) is an interface that connects to a phone line. They supply your PBX with access to the public telephone network. FXO interfaces use FXS signaling. FXS interfaces are what allow you to hook

telephones to your PBX, and FXO interfaces allow you to connect your PBX to real analog phone lines.

5.1.1 Make call via PSTN trunk

For our demo, we want all inside extensions dial '9' through the Analog Trunk callout. In Outbound Routes → add a new Outbound Route as below.

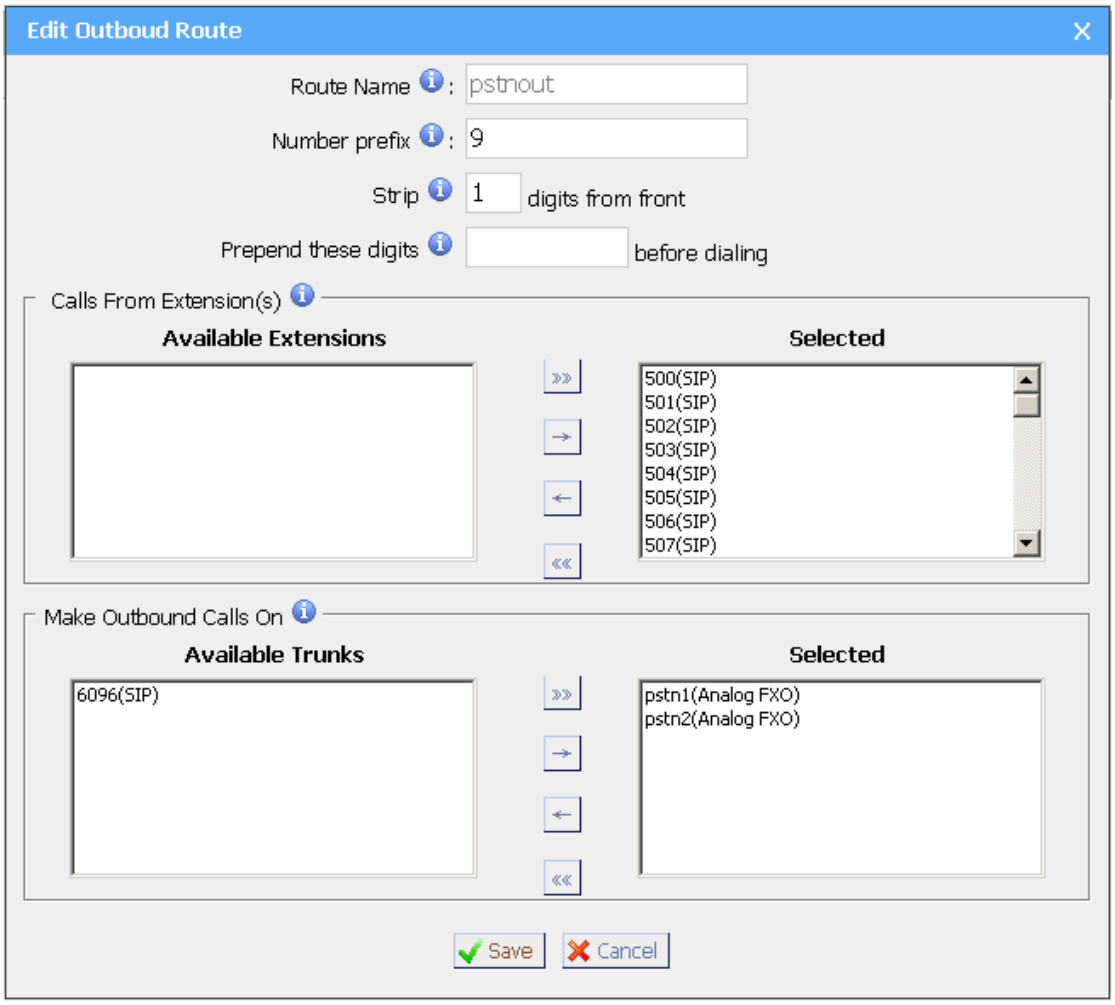


Figure 5-1

As we can see from the Outbound Route of 'pstnout', all numbers start with 9 will be cut the first digit ('9') and send to PSTN (port 1-2).

After we have done above, in the extension we can dial 9 + local number to dial out via PSTN line.

Note: Set number prefix to wild card X and set Strip 0 digits from front will match all the outbound call to go through this outbound route.

5.1.2 Make call via VoIP trunk

For our demo, we want all inside extensions dial '0' throw the VoIP service provider Trunk callout.

1. Add VoIP service provider

Before we do add this, please make sure you have a VoIP service provider account.

Trunks→ SIP Trunk → Add SIP Trunk

Enter your account on this page, and click save.

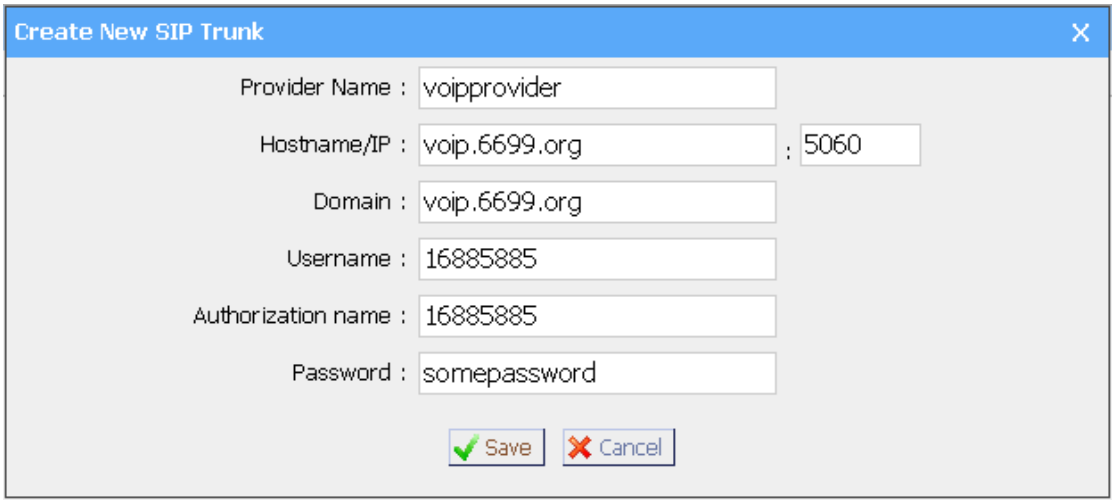


Figure 5-2

2. Add Outbound Routes

As we can see from the Outbound Route of 'voipout', all numbers start with 0 will be cut the first digit ('0') and sent to sip service provide VoIP.

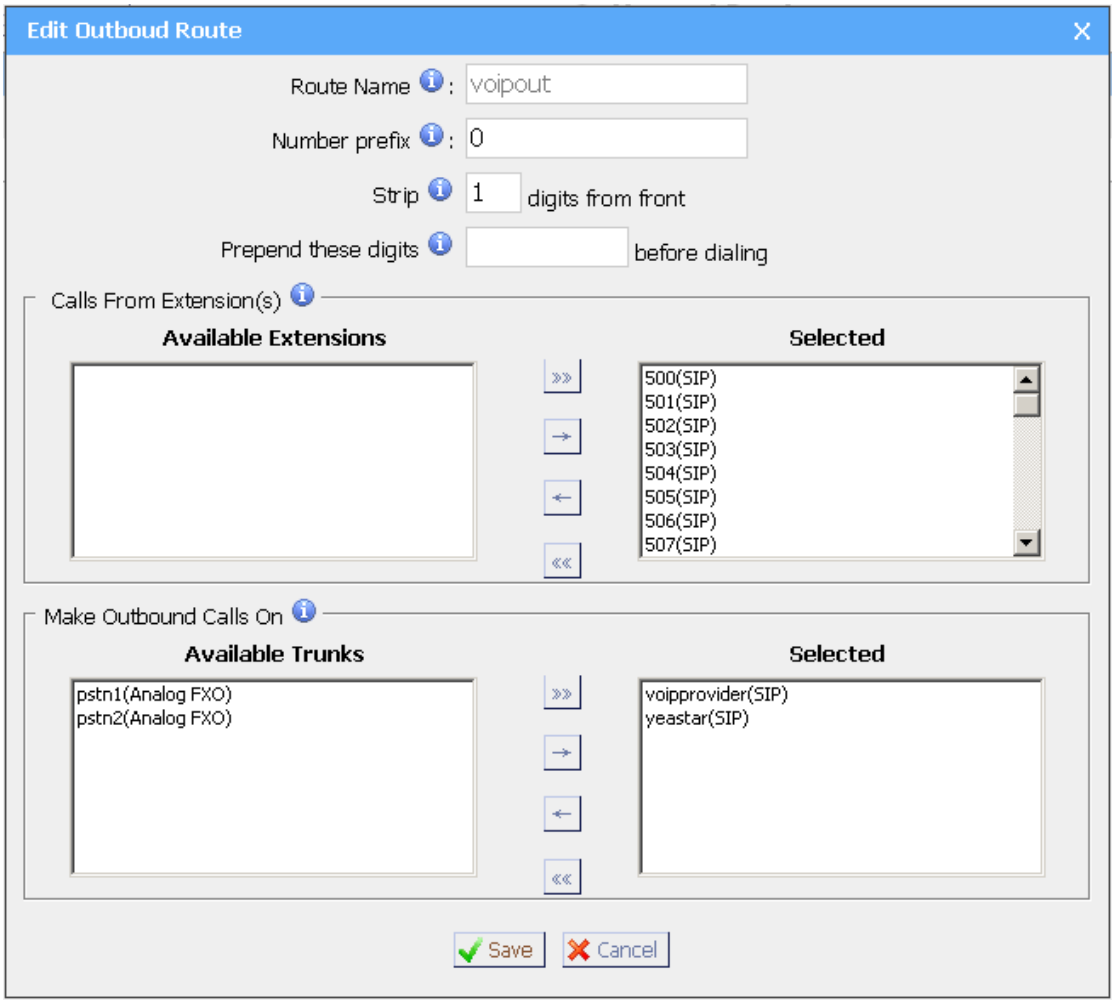


Figure 5-3

So far when we have added two calling rules, any call start with 9 will be route to PSTN, and call starts with 0 will be route to voipprovider.

5.2 Incoming call

5.2.1 Route incoming route to IVR

For our demo, we want all incoming call route to the IVR, press '0' route the call to extension 500, press '1' route the call to extension 501.

1. Add IVR

IVR→ Create New IVR

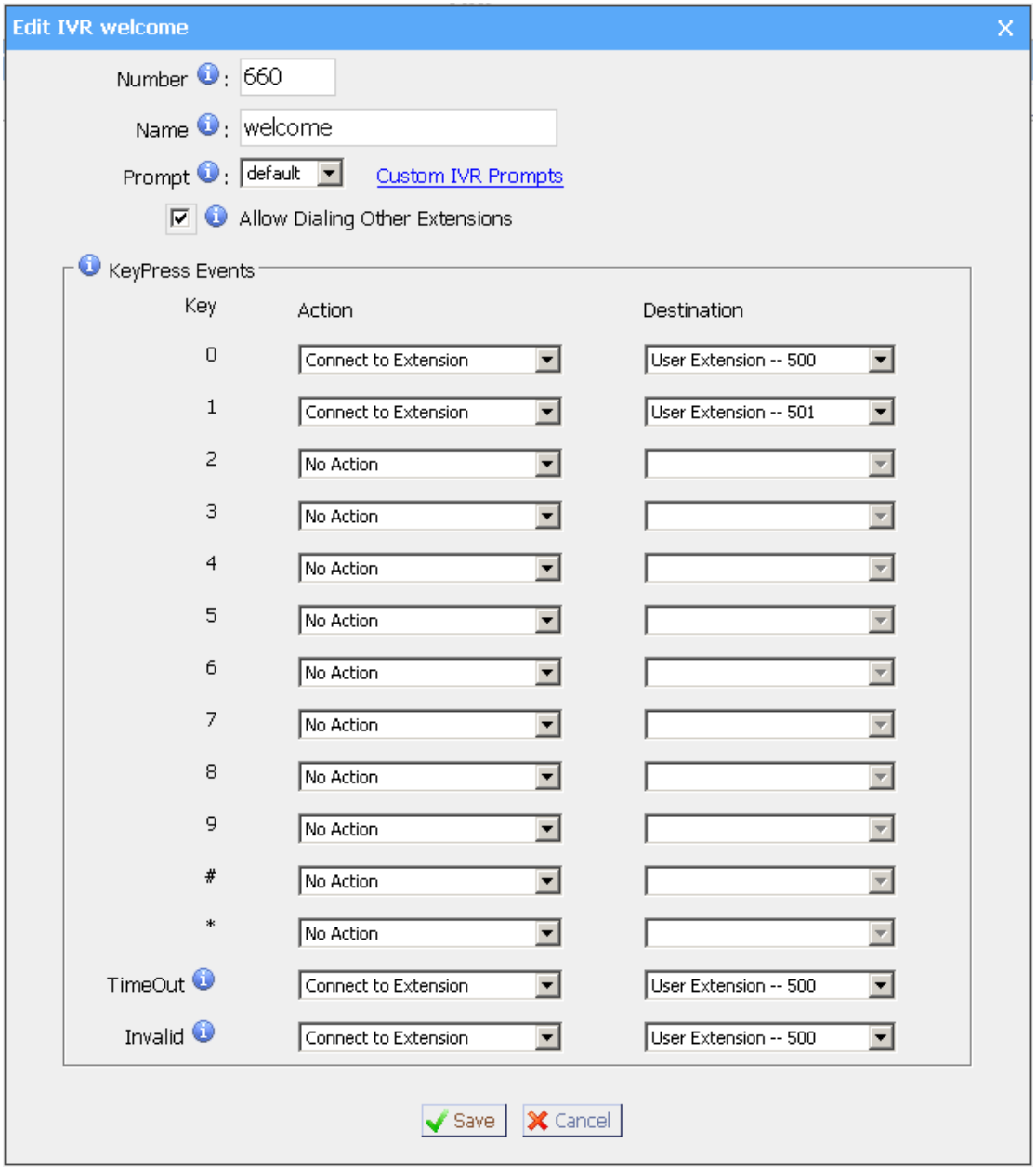


Figure 5-4

2. Add Inbound Routes

As we can see from the Inbound Route of 'allin', all incoming call will be sent to the IVR.

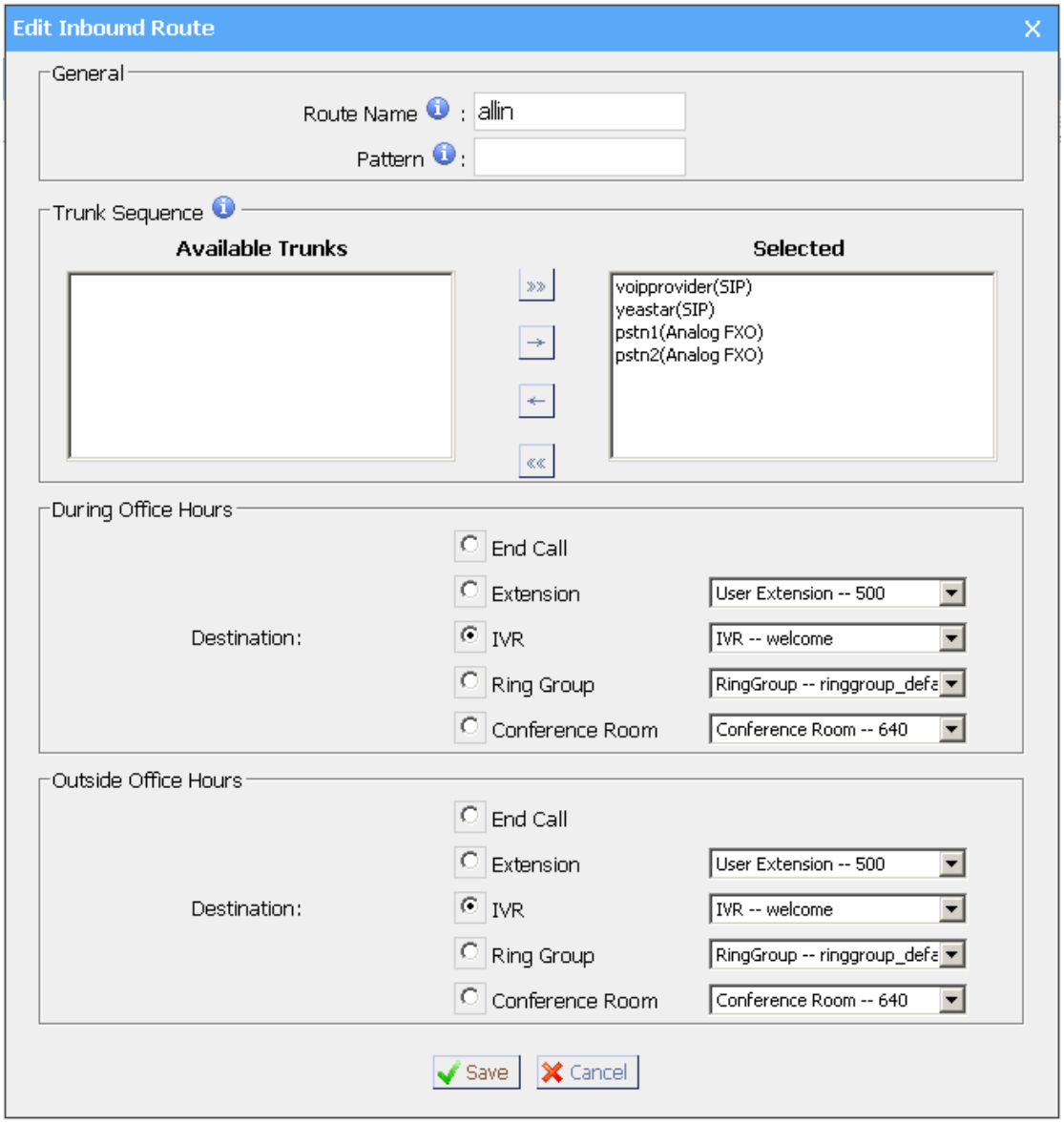


Figure 5-5

APPENDIX A FAQ

Q1. How to Register SIP device?

A1:

1) Register SIP soft phone

Download the x-lite softphone from counterpath website www.counterpath.com

After install the x-lite, right click the panel and select the SIP Account setting and then configure it.

Display Name: 500

User Name: 500

Password: 500

Authorization Name: 500

Domain: 192.168.5.150

2) Register IP Phone (for example, Yealink's T28 IP Phone)

a) Connect the T28's WAN port to the switch. And it can get the IP from your route.

b) Press the 'OK' key on T28 to get the IP of T28.

c) Put the IP on web browser then you can enter the T28 configure page through this IP.

d) Put the SIP extensions info on the T28 IP phones.

Display Name: 501

User Name: 501

Register Name: 501

Password: 501

SIP Server: 192.168.5.150

Use the same method register another T28 to other extension.

Q2. How do I reset MyPBX back to the factory default settings.

A2: To perform a reset, please follow steps below:

Step 1: Press the 'Reset' button on the back of the unit labeled reset.

Step 2: Hold press down for 5 seconds and Watch the lights on the front of the MyPBX. When the status light begins to blink, let go of the reset button.

Step 3: When the status light stops blinking the MyPBX will be back to its factory settings.

Step 4: To access the configuration page, release and renew the IP address on your computer and then open Internet Explorer and type 192.168.5.150 in the address bar.

Step 5: Login to the device with the Username 'admin' and the Password 'password', in order to reconfigure the device.

<Finish>